TROUBLESHOOTING GUIDE	No. TG00069	Ed. 07
OmniPCX Enterprise	Nb of pages :141	Date : 9th January 2013

SUBJECT : Session Initiation Protcol (SIP)

CONTENTS

1. INTRODUCTION 6
2. DOCUMENT HISTORY 6
3. REFERENCES 6
4. ABBREVIATIONS AND NOTATIONS 6
   4.1 Abbrevations 6
   4.2 Notations 6
5. PROTOCOL 7
   5.1 SIP Overview 7
   5.2 SIP Terminology 7
   5.3 SIP structure 8
   5.4 SIP Messages 8
   5.5 SIP Transaction, Dialog & Session 9
      5.5.1 Transaction 9
      5.5.2 Dialog 10
      5.5.3 Session 10
   5.6 SIP Addressing 10
6. SIP LICENSING 11
7. SIP OXE IMPLEMENTATION 12
   7.1 RFCs implemented on OXE 12
      7.1.1 SIP 12
      7.1.2 RTP, T38 & DTMF (used for SIP) 13

7.2 SIPMOTOR processes 14
7.3 OXE duplication 14
7.4 The OXE contains the following compoments: 15
7.4.1 Registrar 15
7.4.2 Proxy 15
7.4.3 Gateway 17
7.4.4 Dictionnary 17
7.4.5 SIP users 17
7.4.6 SIP External Voice Mail 18
7.5 Overview of Interactions between Components 18
7.6 Network number rules 19
7.7 SIP parameters explanation / under the object SIP: 19
7.7.1 SIP Trunk Group 19
7.7.2 The local SIP gateway 21
7.7.3 The external SIP gateways 22
7.7.4 Timer usage for SIP Trunking (Trunk Categoy, by default 31) 24
7.7.5 The SIP proxy 24
7.7.6 SIP Registrar 25
7.7.7 SIP Dictionnary 25
7.7.8 SIP Authentication 26
7.7.9 Quarantined IP Addresses 26
7.7.10 Trusted IP Addresses 26
7.7.11 SIP To CH Error Mapping 26
7.7.12 CH To SIP Error Mapping 28
7.8 SIP parameters explanation / under the object USERS: 28
7.8.1 SIP Device 28
7.8.2 SIP Extension (or SEPLOS) 29
7.9 SIP parameters explanation / under the object SIP Extension: 29
7.10 SIP parameter explanation / under the object External Voice Mail: 30
7.11 SIP parameters explanation / under the object System: 31
8. IP DOMAINS, CODECS AND PCS 32
8.1 IP domains rules 32
8.2 System law for PCM codec 32
8.3 Codecs on SDP 32
8.3.1 Initial offer : the offer sent in an initial INVITE 32
8.3.1 Initial answer : the answer to an initial offer on incoming call 33
8.4 PCS 33

9. CONTENTS OF A SIP MESSAGES (GENERAL VIEW) 34
9.1 The HEADER 34
9.2 The BODY 36
10. EXAMPLES OF COMMON SIP FLOWS 38
10.1 Registration 38
10.2 De-registration 41
10.3 Simple call establishement 42
11. TROUBLESHOOTING 46
11.1 SIPMOTOR processes 46
11.2 SIPMOTOR memory used 47
11.3 Check the SYSTEM and SIPMOTOR backtraces/alarms 48
11.3.1 Backtraces 48
11.3.2 Alarms 49
11.4 SIP traces 51
11.4.1 SIPMOTOR traces 51
11.4.2 Call Handling traces 53
11.4.3 Tcpdump / Network traces 54
11.5 Mantenance commands 55
11.5.1 sip 55
11.5.2 trkstat 55
11.5.3 trkvisu 56
11.5.4 sipaccess 57
11.5.5 sipgateway 57
11.5.6 sipdump 58
11.5.7 sipextgw 67
11.5.8 sippool 68
11.5.9 sipdict 69
11.5.10 sipauth 70
11.5.11 sipregister 71
11.5.12 csipsets 72
11.5.13 csipview com 73
11.5.14 csiprestart 74
11.5.15 sipextusers (Only in R10.x for Open Touch) 74
11.6 Link between SIPMOTOR traces and Call Handling traces 75
11.6.1 Call Handling / SIPMOTOR links implementation 75

11.6.2 General view 76
11.6.3 "neqt" link between SIPMOTOR and Call Handling traces 77
11.7 Information on the SIPMOTOR traces 78
11.8 Follow a call on the SIPMOTOR trace 78
11.9 Traces analyses 81
11.9.1 Incoming SIP call using a SIP Trunk Group: SIPMOTOR point of view 81
11.9.2 Incoming SIP call using a SIP Trunk Group: Call Handling point of view 91
11.9.3 Incoming SIP call in case of SIP extension: SIPMOTOR point of view 97
11.9.4 Incoming SIP call in case of SIP extension: Call Handling point of view 108
11.10 Main call flows explanation 115
11.10.1 Forwards 115
11.10.2 Transfer 117
11.10.3 UPDATE on Early Media 121
11.11 Configuration issues 123
11.11.1 SIP configuration rule 123
11.11.2 SIP alarms generated on OXE 124
11.11.3 Common SIP issues 126
11.11.4 SIP Device issues 130
11.11.5 SIP extension issues 131
11.11.6 SIP External Gateway Issue 132
11.12 Use case 133
11.12.1 Outgoing Call – Cancel sent by OXE after 180 w SDP 133
11.12.2 Telephone-event are not provided on SDP offer 133
11.12.3 Loss of communication with SIP External Voicemail 133
11.12.4 Impossible to let a message when routing via SIP Automated Attendant 133
11.12.5 When call is transfer from a Third Party Server, after few seconds, a Re-Invite is sent by OXE to reroute RTP to a GD card 133
11.12.6 Incoming call from a SIP Third Party Server is rejected by OXE with a SIP Error 488 Not Acceptable Here 133
11.12.7 Incoming call is not recognized as INTERNATIONAL 134
11.12.8 When we attempt to register on SIP External Gateway, OXE answers by a SIP error "482 Loop Detected" 134
11.12.9 When we attempt to register our SIP External Gateway with an external SIP Proxy, SIP Proxy answers by a SIP error "416 Unsupported URI Scheme" 135
11.12.10 Incoming call doesn't transit via Trunk Group configured on SIP Ext Gw 135
11.12.11 Wrong caller number sent in case of forward 136
11.12.12 Diversion/History-Info header is not present 136
11.12.13 SIP-Trunking Name is displayed on calling phone set when call is established 137

11.12.14 From header has not the national format 137
11.12.15 Incoming and outgoing fax communications impossible through SIP Gw137
11.12.16 No Re-Invite with T38 offer sent by OXE 137
11.12.17 External call with secret identity over SIP Provider fails 137
11.12.18 On SIP outgoing call, dynamic ports are used instead of port 5060 138
11.12.19 A "+" character is added on calling number when ISDN call is routed to SIP 138
11.12.20 Diversion Field has not the canonical form 138
11.12.21 Leg1 and leg2 are external set, when OXE user performs a blind transfer, it doesn't work 139
11.12.22 SingleStep Transfer with REFER, no referred-by in the following INVITE 139
11.13 Summary for SIP issue analyse 140
BEFORE CALLING ALCATEL-LUCENT'S SUPPORT CENTER 141
NOTE 141

1. INTRODUCTION

This Troubleshooting Guide deals with SIP (Session Initiation Protocol) and its implementation in OmniPCX Enterprise (OXE), which allows the OXE to connect to SIP phones, SIP trunks and SIP applications like external Voicemail.

The goal is of this document is to explain the functioning of the SIP, to facilitate the troubleshooting and resolution of issues related to SIP

2. DOCUMENT HISTORY

- Ed01: first edition
- Ed02: add “Traces analyses” chapter
- Ed03: add “Use Case” chapter and update 7.11 section
- Ed04: update “SIP Device issues” chapter
- Ed05: update “Use Case” chapter
- Ed06: update 7.7.3 chapter, add new chaper “Timer Usage for SIP Trunking”
- Ed07: add Restriction on “Support of Re-Invite wo SDP”, see 7.7.3 chapter

3. REFERENCES

- OmniPCX Enterprise Technical Documentation

4. ABBREVIATIONS AND NOTATIONS

4.1 Abbrevations

- OXE : OmniPCX Enterprise
- SIP : Session Initiation Protocol
- URI : Uniform Resource Identifier

4.2 Notations

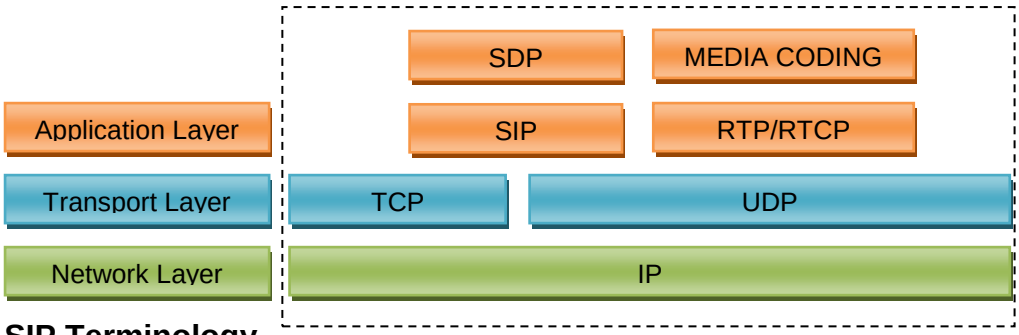
We suggest to pay attention to this symbol, which indicates some possible risks or gives important information.

5. PROTOCOL

5.1 SIP Overview

The SIP protocol is designed to establish, to maintain and to end multimedia sessions between different parties. This protocol is based on the HTTP 1.1

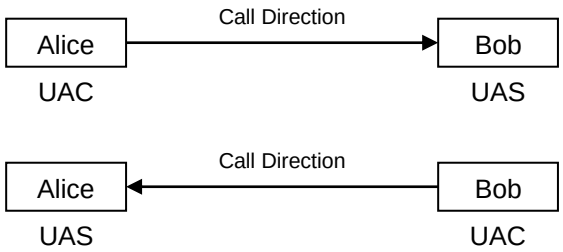
SIP does not provide an integrated communication system. SIP is only in charge of initiating a dialog between interlocutors and of negotiating communication parameters, in particular those concerning the media involved (audio, video). Media characteristics are described by the Session Description Protocol (SDP). SIP uses the other standard communication protocols on IP: for example, for voice channels on IP, Real-time Transport Protocol (RTP) and Real-time Transport Control Protocol (RTCP). In turn, RTP uses G7xx audio codecs for voice coding and compression.



5.2 SIP Terminology

- User Agent (UA)
  - User Agent Client (UAC): Initiator of the SIP requests
  - User Agent Server (UAS): Receiver of the SIP requests (end point)

A SIP equipment can be UAC or UAS according to the direction of the call



- Registrar: A registrar is a server that accepts REGISTER requests and places the information it receives in those requests into the location service for the domain it handles.
  The OmniPCX Enterprise incorporates the function of registrar.
- Location Service: A location service is used by a SIP redirect or proxy server to obtain information about a callee's possible location(s). It contains a list of bindings of address-of-record keys to zero or more contact addresses.
  The OmniPCX Enterprise incorporates the function of location service.
- Proxy, Proxy Server: An intermediary entity that acts as both a server and a client for the purpose of making requests on behalf of other clients. A proxy server primarily plays the role of routing, which means its job is to ensure that a request is sent to another entity "closer" to the targeted user.

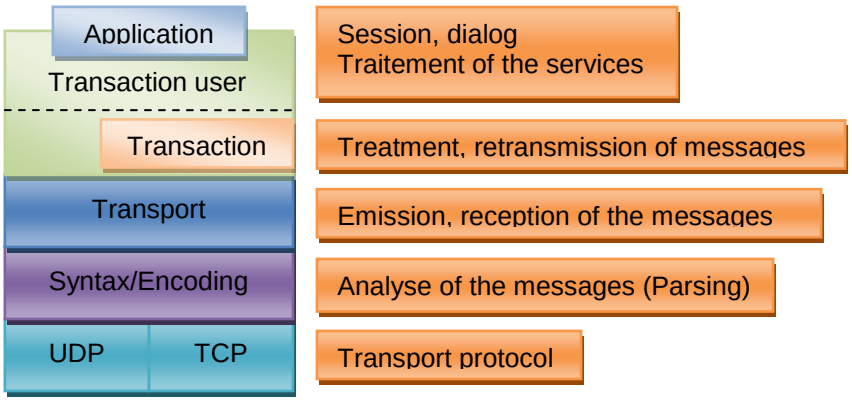
Proxies are also useful for enforcing policy (for example, making sure a user is allowed to make a call). A proxy interprets, and, if necessary, rewrites specific parts of a request message before forwarding it. The SIP proxy is the central actor and first contact for any SIP end user device that wants to initiate a request.

Note: In the OmniPCX Enterprise, the logical functions of registrar, location service and proxy server are co-located and running on the OmniPCX Enterprise call server (CPU/CS/AS) board. The OmniPCX Enterprise proxy server is stateful (it remembers transaction state), call-stateful (stays in the signaling path) and forking (it can redirect requests to multiple destinations).
The name of the SIP domain handled by an OXE node is its node name concatenated with the DNS local domain name defined in SIP/SIP gateway. The main IP address can be substituted wherever appropriate.

- Redirect Server: Provides the client with information about the next hop or hops that a message should take and then the client contacts the next hop server or UAS directly.
  OmniPCX Enterprise does NOT provide a redirect server.
- Gateway: A gateway is a SIP user agent that provides a bridging function between the SIP world and other signaling and telephony systems.

5.3 SIP structure

- The SIP is based on the RFC 3261 (previous RFC 2543), its implementation is next:



5.4 SIP Messages

The main types of requests are:

- REGISTER: message sent by an agent to indicate his current address. This information can be stored in the location server and is used for call routing.
- INVITE: message sent systematically by the client for any connection request.
- ACK: message sent by the client to confirm (acknowledge) the connection request.
- BYE: terminates a call, RTP packet exchange is stopped.
- CANCEL: terminates a call currently being set up.
- SUBSCRIBE - NOTIFY: message used to subscribe to/notify an event (for example: new voicemail message).

- REFER: message requesting an agent to call an address (used for transfers).
- UPDATE: message sent to change the SDP information in early dialog or confirmed dialog.
- MESSAGE: message used to send a message.
- OPTIONS: Requests information about the capabilities of a caller, without setting up a call. Also used for supervision purpose between two Uas.
- PRACK: (Provisional Response Acknowledgement): PRACK improves network reliability by adding an acknowledgement system to the provisional Responses (1xx). PRACK is sent in response to provisional response (1xx).

The remote endpoint answers with a response of one of the following types (main messages answered by OXE):

- 1xx: informational (transaction in progress).
  - The 100 Tyring is particular regarding the other informational answers, used to avoid retransmission of INVITE.
  - The 180 Ringing is used for ring back tone (RBT).
  - The 183 Progress is used to broadcast voice guides.
- 2xx: success (transaction completed successfully).
  - 200 Ok indicates the request was successfull
  - 202 Accepted indicates that the request has been accepted for processing, but the processing has not been completed
- 3xx: forward (the transaction is terminated and prompts the user to try again in other conditions).
  - 301 Moved Permanently
  - 302 Moved Temporarily
- 4xx: The request contains bad syntax or cannot be fulfilled at the server.
- 5xx: The server failed to fulfill an apparently valid request
- 6xx: The request cannot be fulfilled at any server

Regarding the unsuccessfull answers, for signification, use the RFC 3261.

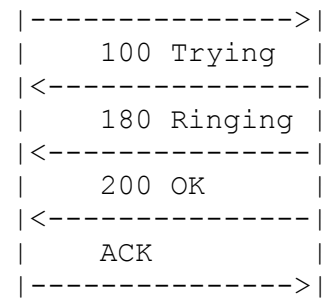
5.5 SIP Transaction, Dialog & Session

5.5.1 Transaction

The transactions have to separated:

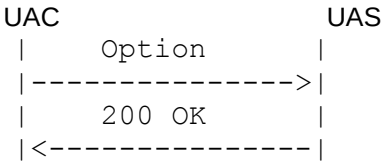
- The INVITE transaction
  - The INVITE transaction is composed of three ways
    - INVITE sends from the client to the server
    - Answers send from the server to the client
    - Client must send an ACK
  - If these three steps are respected, a INVITE transaction is done
  - Example

```
UAC                    UAS
 |      INVITE         |
```



An INVITE transaction (with all the information from this INVITE) can be called a “leg”.

- The Non-INVITE transaction
  - The Non-INVITE transaction is composed of two ways
    - Request sends from the client to the server
    - Answers send from the server to the client
    - No ACK
  - If these three steps are respected, a Non-INVITE transaction is done
  - Example



5.5.2 Dialog

Dialogs are created through the generation of non-failure responses. When an INVITE is answered with a 200 Ok, the dialog is opened.

- A dialog is identified by :
  - a call identifier
  - a local tag
  - a remote tag

5.5.3 Session

A session is open for audio or video exchanges. The UAC and UAS receives the information to open a RTP flow, in that case, the session is opened.

5.6 SIP Addressing

SIP entities are identified using SIP URIs (Uniform Resource Identifier). A SIP URI is of the form of sip:username@host, similar to an email address, typically containing a username and a host name delimited by @ (at) character. The host part can be an IP address, the name of a machine, or a Fully Qualified Domain Name (FQDN), i.e. the name of a domain. The username part can be a telephone number.

- Examples for SIP URIs:

  sip:alice@atlanta.com

  sip:1212@gateway.com

  sip:alice@192.0.2.4

In OmniPCX Enterprise, the more specific term URL (Uniform Resource Locator) is generally used instead of URI, since OXE is more concerned about location aspects rather than identification aspects.

For OXE uses on the username part numbers and no names.

6. SIP LICENSING

Here the next licenses for SIP (under spadmin):

```
177 M SIP users                          =      13/ 25
...
185   SIP Gateway                        =          1
...
188   SIP network links                  =         45
...
345 M SIP extension users                =       8/ 25
```

- The license 177 corresponds to the maximum number of SIP users (SIP Extension & SIP Device).
- The license 185 corresponds to the use of the SIP on the OXE (activation).
- The license 188 corresponds to the maximum number of SIP Calls available all the SIP elements (SIP calls thru Trunk group and SIP extension).
- The license 245 corresponds to the maximum number of SIP Extension users.

Another information link to SIP is important, the PARAMAO 3 used for the creation of the SIP Trunk Group (under cfgUpdate):

```
  5 Trunks                                :     5000
```

This value is calculated according to the number of Trunk Groups managed via ACTIS (including SIP).

7. SIP OXE IMPLEMENTATION

7.1 RFCs implemented on OXE

7.1.1 SIP

- RFC 2543 (obsolete by RFC 3261,3262, 3263,3264, 3265): SIP: Session Initiation Protocol
- RFC 2782: A DNS RR for specifying the location of services (DNS SRV)
- RFC 2822: Internet Message Format
- RFC 3261: SIP: Session Initiation Protocol
- RFC 3262: Reliability of Provisional Responses in SIP (PRACK)
- RFC 3263: SIP: Locating SIP Servers
- RFC 3264: An Offer / Answer model with SDP
- RFC 3265: SIP-Specific Event Notification
- RFC 3311: The SIP UPDATE Method (session timer only)
- RFC 3323: Privacy Mechanism for the Session Initiation Protocol (SIP)
- RFC 3324: Short term requirements for network asserted identity
- RFC 3325:Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks
- RFC 3265: SIP-specific Event Notification
- RFC 3515: The Session Initiation Protocol (SIP) Refer method
- RFC 3891/3892: The Session Initiation Protocol (SIP) 'Replaces' Header/ Referred-By Mechanism
- RFC 3398: Integrated Services Digital Network (ISDN) User Part (ISUP) to SIP Mapping
- RFC 3966: The telephone URI for telephone numbers (url tel not supported)
- RFC 4497: Inter-working between SIP and QSIG
- RFC 5373: Requesting Answering Modes for the Session Initiation Protocol
- RFC 4244: An Extension to the Session Initiation Protocol (SIP)for Request History Information
- RFC 3326: The Reason Header Field for the Session Initiation Protocol (SIP)
- RFC 3428: Session Initiation Protocol (SIP) Extension for Instant Messaging (partial)
- RFC 3608: Service Route header
- RFC 3327: Path Header
- RFC 2246: The TLS Protocol Version 1.0
- RFC 3268: Advanced Encryption Standard (AES) Cipher suites for Transport Layer Security (TLS)
- RFC 3280/5280: Internet X.509 Public Key Infrastructure Certificate and Certificate Revocation List (CRL) Profile
- RFC 3711: The Secure Real-time Transport Protocol (SRTP) (media integrity)
- RFC 4568: Session Description Protocol (SDP) Security Descriptions for Media Streams
- RFC 5806: Diversion Indication in SIP

7.1.2 RTP, T38 & DTMF (used for SIP)

- RFC 2617: HTTP Authentication : Basic and Digest Access Authentication
- RFC 1321: Authentication for Outgoing calls
- RFC 2833/4733: DTMF Transparency. RFC 2833 replaced by RFC 4733
- RFC 3842: A message Summary and Message Waiting Indication Event Package
- RFC 4028: The session timers in the Session Initiation Protocol
- RFC 3725: Best current practices for Third party Call Control (3 pcc) in SIP (scenario 1). Invite without SDP.
- RFC 3960: Early Media (partial): Gateway model not supported
- RFC 1889/1890: RTP : A transport protocol for Real-Time applications
- RFC 2198: RTP Payload for Redundant Audio data
- RFC 3550: RTP: A Transport Protocol for Real-Time application (audio only)
- RFC 3551: RTP Profile for Audio and Video Conferences with Minimal Control (audio only)
- RFC 3711: The Secure Real Time. Supported on A-LU IP Phone and Softphone
- RFC 3362: T38 ITU-T Procedures for real time Group3 Fax Relay / communications over IP

7.2 SIPMOTOR processes

In the OmniPCX Enterprise, the logical functions of registrar, location service, proxy server and gateway are co-located in the process called **sipmotor**, running on the CPU7/CS2/AS board.

You may use the linux **ps** command to verify that the SIP processes are running :

Example in R9.1:

```
(1)OXE> ps -edf | grep sip
root      2247   820  0 Jan05 ?        00:00:00 /DHS3bin/servers/sipmotor
root      2248  2247  0 Jan05 ?        00:00:31 /DHS3bin/servers/sipmotor
root      2249  2247  0 Jan05 ?        00:00:00 /DHS3bin/servers/sipmotor
root      2250  2247  0 Jan05 ?        00:00:00 /DHS3bin/servers/sipmotor
root      2251  2247  0 Jan05 ?        00:00:00 /DHS3bin/servers/sipmotor
```

Example in R10.0:

```
(1)OXE> ps -edf | grep sip
root      2202   801  0  2011 ?        00:00:00 [#sipmotor]
root      2203  2202  0  2011 ?        00:00:00 [sipmotor_tcl]
root      2204  2202  0  2011 ?        00:00:00 [sipmotor]
root      2205  2202  0  2011 ?        00:00:00 [sipmotor_dump]
root      2206  2202  0  2011 ?        00:00:00 [sipmotor_presen]
```

All processes can be forced to reset with the command:

- **dhs3_init -R SIPMOTOR**, this command stops properly the SIPMOTOR processes and restarts them.

```
(1)OXE> dhs3_init -R SIPMOTOR
```

They will be automatically relaunched after a few seconds.

The next commands can be used as well:

- **killall sipmotor**, this command kills the SIPMOTOR processes and restarts them.
- **kill -9 "father pid"**, this command kills the SIPMOTOR processes and restarts them.

If no licenses about SIP are present, the SIPMOTOR processes are not running.

7.3 OXE duplication

In case of OXE duplication, the SIPMOTOR is complety started on the Stand-By CPU, but acting as Stand-By (cannot treat the SIP requests). The Main CPU puts up to date the Stand-By CPU about the SIP contexts (Calls, registrations, subscriptions, etc...). In case of bascul, the SIP phone calls are maintained and the registration and subscriptions are kept.

In Case of spatial redundancy with dual subnetworks (2 main IP addresses), the SIP is using the FQDN of the OXE (nodename + DNS local domain name) for the SIP messages and also for the responses of the SIP messages, in that case, the remote SIP equipment must be use it. A use of external DNS server is recommended to resolve this FQDN.

7.4 The OXE contains the following compoments:

7.4.1 Registrar

Register the addresses of the SIP terminals (“Location Service”)

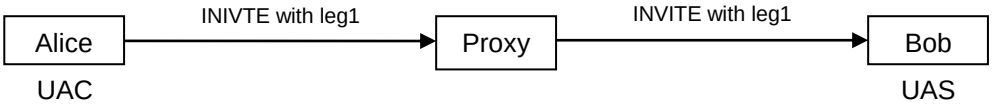
- The REGISTRAR is containing in the file “**localize.sip**” under /tmpd. If for any reasons you need to clear all entries in the registrar database, remove this file and then restart the SIPMOTOR:

```
(1)OXE> rm /tmpd/localize.sip
(1)OXE> dhs3_init -R SIPMOTOR
```

7.4.2 Proxy

Entity between the Client and the Server, the proxy is used to route the SIP requests.

- The call can be routed between 2 SIP terminals, for instance Alice calls Bob (both are SIP), in that case, Alice sends a SIP request to the proxy, and the proxy sends this request to Bob.
- The proxy can be used only for the authentication of the SIP equipment for Registration or SIP request.
    - The proxy can modify the request by adding information like a Via, Record-route, etc...



The INVITE is the same on each proxy sides, to get this behavior, and the UAC manage the IP address of the OXE SIP proxy as the “Outbound proxy”

Here an example:

- The UAC IP address: 172.27.143.184
- The proxy IP address: 172.27.143.186
- The UAS FQDN: oxe-ov.alcatel.fr (IP address: 172.27.141.151)

```
Fri Jun 29 14:08:10 2012 RECEIVE MESSAGE FROM NETWORK (172.27.143.184:5060 [UDP])
----------------------utf8-----------------------
INVITE sip:172.27.143.186 SIP/2.0
Via: SIP/2.0/UDP 172.27.143.184:5060;rport;branch=z9hG4bKPjX7-GJh79mg04nEbZ0yxYsWP3MCiy4C4H
Max-Forwards: 70
From: <sip:31027@oxe-ov.alcatel.fr>;tag=BJ2er-g.ONc2M.MQJ9qO.wfpLyp8qfQ3
To: <sip:31002@oxe-ov.alcatel.fr>
Contact: <sip:31027@172.27.143.184:5060>
Call-ID: L9TrfBGqqYwgo6CR.c9YtaiyulB9OGVU
CSeq: 23308 INVITE
Route: <sip:oxe-ov.alcatel.fr;transport=udp;lr>
Route: <sip:31002@oxe-ov.alcatel.fr;transport=udp>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE, OPTIONS
Supported: 100rel, norefersub
User-Agent: OmniTouch 1.5.13.7
Content-Type: application/sdp
Content-Length:   283
```

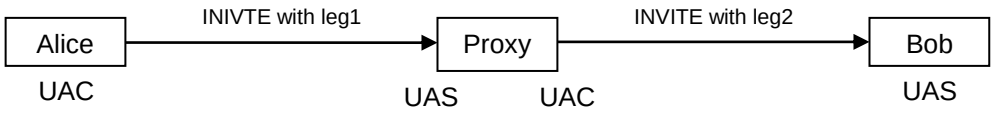
The OXE SIP proxy receives an INIVTE with the information “Route” corresponding to the final end point for the SIP call. In that case, the OXE SIP proxy acts like a proxy (not a back to back). Due to this, the proxy sends the next INVITE to the final SIP endpoint.

```
Fri Jun 29 14:08:10 2012 SEND MESSAGE TO NETWORK (172.27.141.151:5060 [UDP]) (BUFF LEN = 1130)
----------------------utf8-----------------------
INVITE sip:31002@oxe-ov.alcatel.fr;transport=udp SIP/2.0
Route: <sip:oxe-ov.alcatel.fr;transport=udp;lr>
Record-Route: <sip:172.27.143.186;lr;transport=UDP>
Via: SIP/2.0/UDP 172.27.143.186;branch=z9hG4bK1053e27e7fdda06c573798bc91cd12a29c49e03527107ccdabde727c92e5b987
Via: SIP/2.0/UDP 172.27.143.184:5060;received=172.27.143.184;rport=5060;branch=z9hG4bKPjX7-
GJh79mg04nEbZ0yxYsWP3MCiy4C4H
Max-Forwards: 69
From: <sip:31027@oxe-ov.alcatel.fr>;tag=BJ2er-g.ONc2M.MQJ9qO.wfpLyp8qfQ3
To: <sip:31002@oxe-ov.alcatel.fr>
Contact: <sip:31027@172.27.143.184:5060>
Call-ID: L9TrfBGqqYwgo6CR.c9YtaiyulB9OGVU
CSeq: 23308 INVITE
Allow: PRACK,INVITE,ACK,BYE,CANCEL,UPDATE,SUBSCRIBE,NOTIFY,REFER,MESSAGE,OPTIONS
Supported: 100rel,norefersub
User-Agent: OmniTouch 1.5.13.7
Content-Type: application/sdp
Content-Length: 283
Session-Expires: 1800
```

The proxy is adding some information on the INVITE sent to the final SIP end point, but the INVITE is the same than the one received (same Call-ID, same FROM, same TO, same TAGs, etc...)

- The REQUEST-URI has been modified according to the information from the Route from the first INVITE.
  - INVITE sip:31002@oxe-ov.alcatel.fr
- Information added:
  - Via: SIP/2.0/UDP 172.27.143.186; branch=z9hG4bK1053e27e7fd…
    - Correponding to the proxy “identification“
  - Record-Route: <sip:172.27.143.186;lr;transport=UDP>
    - Correponding to the path for the answers (the answers must be sent to this IP address)
  - Session-Expires: 1800
    - Corresponding to the session timer used on the proxy

- The Proxy can be used as a Back-to-Back, in that case, on each side, two different legs will be found



Two different INVITEs on each proxy sides.

There are no specific information on the INVITE, because the proxy is acting as an UAS for the caller and an UAC for the called party.

7.4.3 Gateway

Entity between SIP world to legacy world, the gateway is used to establish a call from a SIP equipment to an ISDN link, to a legacy set, etc… and vice versa

- Do not confuse the SIP gateway with the OmniPCX Enterprise media gateway boards. The SIP gateway is a logical entity that resides within the call server (CS) and is responsible for the SIP signaling for the conversation setup, while the media gateway boards (GD, GA, INTIP) are the physical devices where the media session will be established when calling to a classic PBX set.

- There is one and only one internal SIP gateway. But there can be many different external SIP gateways (we will come back to this in a later section).

- The SIP gateway is associated to a SIP trunk group. Although there can be many SIP Trunk Groups, there is only one SIP trunk group which is associated to the local SIP gateway. We call this special trunk group the local SIP trunk group.

7.4.4 Dictionnary

Contains the SIP users created on the OXE, it is the database that holds the mapping between SIP URLs and PBX directory numbers (MCDUs). Each registered SIP terminal is automatically added to the dictionnary. Classic PBX terminals are added only if a SIP URL is defined for them in the user management.

- Most of the time you shouldn't do anything with the Dictionnary. Everything will be handled automatically. You need to access the SIP Dictionnary configuration only for configuration of aliases.

7.4.5 SIP users

On the OXE , we have the two types of SIP users:

- SIP Device
    - The SIP device is considered as an external SIP user, it means that the SIP device is linked to the local SIP gateway, and use its configuration
    - The phone features are limited

- SIP Extension(or SEPLOS)
    - The SIP extension is considered as an internal SIP user, it means that the SIP extension can be access to some OmniPCX Enterprise services and phone features
    - It can used some OmniPCX Enterprise's prefixes, can be declared as a room set, etc…
    - The phone features available depend also of the SIP phone itself.
    - A SIP extension is attached to a virtual UA board, idem as an IPtouch.

On OXE, it is necessary to understand that a SIP extension user is different than the SIP phone associated to this user, for instance:

- If the SIP phone is fowarded, that doesn't mean that the user is forwarded.
- If the user is forwarded, that doesn't mean that the SIP phone is forwarded.

It is very important to reminder this behaviour.

The declaration of a SIP user binds the information configured in the SIP set with the information stored into the database of the OmniPCX Enterprise.

If you don't fill in the SIP part in the OmniPCX Enterprise user configuration, the default values will be :

- URL User Name = MCDU of the user.
- URL Domain = SIP domain name of the OmniPCX Enterprise, i.e. the SIP set is considered as registered on the OmniPCX Enterprise.

This is usually exactly what we want so you shouldn't modify anything here.
After the creation of the user a corresponding entry will automatically be added to the SIP Dictionnary.

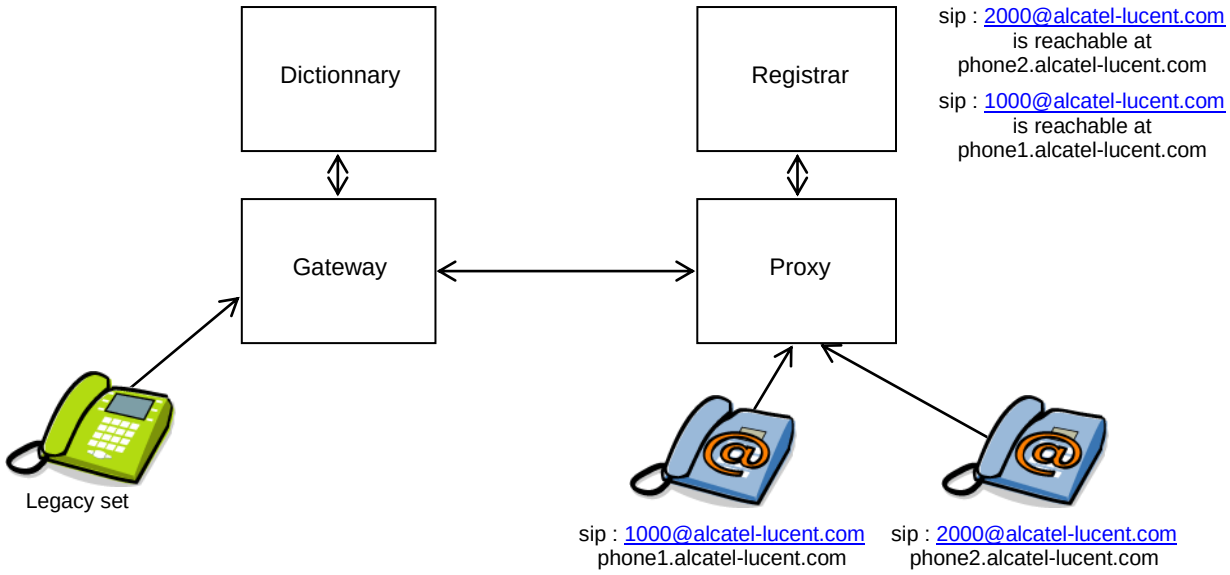
Note: The value for the URL (<username>@<domainname>) configured on the SIP set and in the OmniPCX Enterprise SIP Dictionnary MUST match. This can be an issue if you modified one of these parameters by hand and not the other one.

7.4.6 SIP External Voice Mail

On the OXE, it is possible to connect external voice mail, as the OmniTouch 8440, to be able to manage it and use it, the local SIP gateway must be managed first.

7.5 Overview of Interactions between Components

The following diagram shows the relations between the functional SIP modules in OmniPCX Enterprise :



7.6 Network number rules

The OXE is using network (or subnetwork or routing tables) for different applications and must be unique for each application. It is very important for SIP to respect the next configuration:

- The ABC-F network is using its one network number (managed on System parameter).
- The VPN are using different network numbers according to the configuration.
- The local Hybrid Link (for CCD) is using its one network number.
- For the local SIP gateway, it is necessary to use a network number used only for it, do not use a network number used by another application.
- Each external ABC-F gateways are using their one network number.

These rules must be respected to avoid SIP issues.

7.7 SIP parameters explanation / under the object SIP:

7.7.1 SIP Trunk Group

(idem for R9.1 and R10.x)

The SIP Trunk Group is mandatory if you want to use the Local SIP gateway or an external SIP gateway (not necessary for SEPLOS users).

The Trunk Group is used to give channels for SIP calls, according to its type and configuration, the features available are differents.

- Different types of SIP trunk Groups are available on OXE:
    - The SIP ABCF Trunk Group.
        - Maximum of features available, the number of accesses is from 2 to 32 (31 channels for one access).
    - The SIP ISDN Trunk Group.
        - Less features available compares to ABCF Trunk Group, the number of accesses is from 2 to 32 (31 channels for one access).
    - The Mini SIP ABCF Trunk Group.
        - Maximum of features available, the number of accesses is from 2 to 32 (2 channels for one access).
    - The Mini SIP ISDN Trunk Group.
        - Less features available compares to ABCF Trunk Group, the number of accesses is from 2 to 32 (2 channels for one access).
- Level of service depending on used trunk group :
    - Call transfer
        - ISDN :Using re-INVITE in the opened dialog.
        - ABC-F :Via REFER, « referred-by » and « replaces ».
    - Call forward
        - ISDN :Done internally.
        - ABC-F :Redirecting with 3xx. New call has to be performed by remote party.
    - Call discrimination
        - ISDN :Same as ISDN.
        - ABC-F :No discrimination.

- To create an SIP Trunk Group, go under /Trunk Groups

Trunk Group Type	: Select T2 for all the different types of SIP Trunk Group
Trunk Group Name	: Manage a name for the SIP Trunk Group
Number Compatible With	: Keep "-1" everytime, don't manage another value
Remote Network	: Enter a Remote network number, for an ABCF TG, use the dedicated number, for ISDN TG keep 255 (idem as legacy T2 ISDN Trunk group)
Node number	: Enter the node number of your OXE
Q931 Signal variant	: - For an ABCF SIP Trunk group, select ABC-F - For an ISDN SIP Trunk Group, select ISDN
Number Of Digits To Send	: Keep "0" everytime, don't manage another value
T2 Specification	: - Select SIP for a SIP Trunk Group (ISDN or ABCF) - Select Mini SIP for a Mini SIP Trunk group (ISDN or ABCF)
Public Network COS	: According to the value manage, the OXE will use the rights of the associated category
DID transcoding	: This parameter is set to "True" only in case of ISDN SIP Trunk Group (or Mini SIP ISDN Trunk Group)
Associated Ext SIP gateway	: Enter the external SIP gateway used if there is no DCT managed on the ARS route, the DCT from the ARS route is used in priority (From R10.1)

- To create a SIP Trunk Group, go under /Trunk Groups/Trunk Group

IP Compression Type	: - "Default" means only the system algorithm used on SDP - "G711" means the use of the sytem algorithm and the PCM with the system law
Trunk COS	: According to the value manage, the OXE will use the rights of the associated category
IE External Forward	: Select "Diverting leg information" if you want to use the History-Info or Diversion header (only from R10.x for Diversion header)

- To create an SIP Trunk Group, go under /Trunk Groups/Trunk Group/Virtual accesses for SIP

Number of SIP Accesses	: Enter the number of SIP accesses needed on the SIP TG (value from 2 to 32)

Some other parameters can be modified with Alcatel-lucent's agreement according to the AAPP
tests(applications and phones) and/or the SIP Interoperability tests (SIP providers).

7.7.2 The local SIP gateway

(idem for R9.1 and R10.x)

Used for the local SIP users (SIP Device) and the external Voice mail

- To manage the Local SIP gateway, go under /SIP/SIP Gateway

SIP Subnetwork	: Corresponds to the local SIP network (different than the ABC-F network and used **only** for the local SIP gateway).
SIP Trunk Group	: Corresponds to the SIP Trunk group (better to use an ABCF SIP Trunk group)
IP Address	: Corresponds to the IP address of the CPU (autofill)
Machine name – Host	: Corresponds to the nodename associated to the main IP address (managed via netadmin - autofill).
SIP Proxy Port Number	: Corresponds to the SIP port number (by default 5060).
SIP Subscribe Min Duration	: Corresponds to the minimum duration of a SIP subscription (for message waiting indication or for result of a transfer).
SIP Subscribe Max Duration	: Corresponds to the maximum duration of a SIP subscription (for message waiting indication or for result of a transfer).
Session Timer	: Corresponds to the timer value to supervise an active SIP session. A RE-INVITE or UPDATE message is sent before SIP Session Timer expiry (for all SIP elements).
Min Session Timer	: Corresponds to the mimimum session timer value accepted by the OXE. When a SIP call is established, the session timer is negociated between the two parties.
Session Timer Method	: Corresponds to the method used for session timer, the OXE sends a RE-INVITE or an UPDATE message.
DNS local domain name	: Corresponds to local DNS suffix used for SIP. The FQDN of the OXE is the nodename + this domaine name (mandatory in case of spatial redondancy).
DNS type	: Corresponds to the DNS mode (A or SRV).
SIP DNS1 IP Address	: IP address of the first DNS server. (Not manage the CPU IP address)
SIP DNS2 IP Address	: IP address of the second DNS server. (Not manage the CPU IP address)
SDP in 18x	: Used to put SDP information on th 18x sent by the OXE.
Cac SIP-SIP	: To allow or not, the domains control in SIP to SIP communications.
INFO method for remote extension	: Using the INFO method for DTMF in case for the Nokia Call Connect (NCC) only.
Dynamic Payload type for DTMF	: Payload value used for DTMF, default value 97 (used by the SIP device for instance).

7.7.3 The external SIP gateways

Used to connect external SIP equipments // applications (SIP provider, Call centre application, etc…).

SIP External Gateway ID	: Id of the gateway
Gateway Name	: Name given to the gateway
SIP Remote domain	: IP address or FQDN of the remote SIP equipment (if FQDN, need to use a DNS server)
PCS IP Address	: PCS IP address used to backup this gateway in case of link failure with the CPU
SIP Port Number	: SIP port number used to send SIP messages on the remote gateway
SIP Transport Type	: Transport type for SIP messages (UDP or TCP)
Belonging Domain	: Used to define the domain part of the URI (FROM and PAI) on the SIP message
Registration ID	: Registration id used on the user part if the remote gateway needs it
Registration ID P_Asserted	: Used the registration ID on the P_Asserted Identity (PAI)
Registration timer	: Timer used for registration (0 = no registration)
SIP Outbound Proxy	: Send the messages (INVITE and REGISTER) on this address
Supervision timer	: Used to supervised the remote gateway (OPTION message sent)
Trunk group number	: SIP trunk group used for this SIP gateway
Pool Number	: Can associate 2 external SIP gateways in one pool (Load Balancing)
Outgoing realm	: Realm of the remote gateway (Outgoing messages authentication)
Outgoing username	: Username from the remote gateway (Outgoing messages authentication)
Outgoing Password	: Password from the remote gateway (Outgoing messages authentication)
Incoming username	: Username used by the remote gateway (Incoming messages authentication)
Incoming Password	: Password used by the remote gateway (Incoming messages authentication)
RFC 3325 supported by the distant	: PAI supported for Outgoing calls
DNS type	: DNS requests types (A or SRV)
SIP DNS1 IP Address	: IP address of the first DNS server (Not manage the CPU IP address)
SIP DNS2 IP Address	: IP address of the second DNS server (Not manage the CPU IP address)
SDP in 18x	: Used to put SDP information on th 18x sent by the OXE
Minimal authentication method	: Used to activate or not the authentication (DIGEST or SIP none)
INFO method for remote extension	: Using the INFO method for DTMF in case of remote extension
Send only trunk group algo	: Used to send only the algorithm managed on the SIP TG
To EMS	: Used to activate the RFC4916 (Add specific fields for identification on EMS)
SRTP	: Used in case of SIP TLS to select the RTP mode (secured or not) (From R10.0)
Routing Application	: - False: SDP sets on the SIP messages (INVITE, 200ok...) - True: No SDP on the SIP messages, this parameter is used for some specific configuration for carriers
Ignore inactive/black hole	: Only for SIP ABC-F.

- False means that the receipt of a Re-INVITE, whose SDP indicates either inactive or c=0.0.0.0 is handled as an Hold request.
- True means that the same kind of Re-INVITE leads the RTP flow towards the remote party to be cut.

Contact with IP address : In case of spatial redundancy with dual subnetworks, the IP address of the main Call Server is put on the Contact field instead of the FQDN of the OXE

Dynamic Payload type for DTMF : Corresponds to the payload value for DTMF must be the same than value from the remote SIP equipment.

100 REL for Outbound Calls : - Not supported : Outbound INVITE doesn't indicate 100Rel parameter.
- Supported : Default Value. Outbound INVITE indicates 100Rel in "Supported" header.
- Required : Outbound INVITE indicates 100Rel in "Require" header.

100 REL for Incoming Calls : - Not requested : Default value. 18x response triggered from OXE doesn't indicate 100Rel in "Require" header.
- Required mode1 : 18x response triggered from OXE indicates 100Rel in "Require" header only if it provides SDP.
- Required mode2 : 18x provisional response triggered from OXE indicates 100Rel in "Require" header.

Gateway type : Use to define if the remote SIP gateway is un Open Touch or not, keep default configuratiuon if it is not a Open Touch (From R10.0)

Re-Trans No. for REGISTER/OPTIONS: Number of retransmission of SIP REGISTERs/OPTIONs messages, from 1 to 10 (From R10.0)

P-Asserted-ID in Calling Number : - If True, Calling Number is filled from P-Asserted-ID header
- If False, Calling Number is filled from FROM header.
(From R10.0)

Trusted P-Asserted-ID header : Octet3a_Calling is filled based on this parameter (Used, only when there is P-Asserted-ID header)
(From R10.0)

Diversion Info to provide via : In the Outbound INVITE the selected Header is added to provide information about Call deflection/forward. The OXE can use History-Info (RFC 4244) or Diversion (RFC 5806)
(From R10.0)

Outbound calls only : - if False, the existing procedure applies.
- If True, the External Gateway is skipped during the lookup procedure of the origin of the call. The way to determine the origin of an inbound call, e.g. the External Gateway it comes from, is made in such a way that in that topology, the lowest External Gateway, in term of numbering, is chosen.(From R10.1)

SDP relay on Ext. Call Fwd : In case of SIP trunk to SIP trunk call rerouting (essentially external to external call forward), in order to adapt specific SIP profile, OXE offers the possibility to transit SDP answers received in 180 or 183 on outgoing leg only in 180 answer on incoming leg.
- Default : normal procedure apply. SDP can transit with 183 message depending on call flow.
- 180 only : any SDP received in 180 and 183 on outgoing leg will not transit on incoming leg in 183 provisional answer but only in 180 ringing one.(From R10.1)

Trusted From header : Octet3a_Calling is filled based on this parameter (Used, only when there is no P-Asserted-ID header). To be used when calling number is found in FROM header and should be considered as trusted by the system. (From R10.0)

Support Re-invite without SDP : - if True, the OXE will send a REINVITE without SDP in case of supervised transfer between two SIP calls, only if the SIP equipment support it.
- if False, the OXE will send a REINVITE with SDP. (From R10.1)

Restriction : When PRACK is supported, this parameter must be ckeched at **False**

Proxy identification on IP address : - if True, a dynamic "DNS cache" per SIP External Gateway is handled by OXE to store the IP address(es) to where Register and further INVITE may be sent. At the beginning of the procedure, this DNS cache is empty. (From R10.1)

Registration on proxy discovery : - if True, used when SIP Carrier provide more than one outbound proxy. As soon as, on carrier side a switch happen from one proxy to anthoer, calls cannot be neither delivered to OXE, nor accepted by the carrier as long as a new registration is not triggered by OXE. (From R10.1)

Nonce caching activation : (From R11)

RFC 5009 supported / Outbound call : (From R11)

FAX Procedure Type : (From R11)

Type of codec negotiation : (From R11)

7.7.4 Timer usage for SIP Trunking (Trunk Categoy, by default 31)

This only applies to SIP Trunking Call Handling where generic timers are used

Timer	Value	Meaning
Timer T302	15s	Related to SETUP_ACK
Timer T303	10s	Related to Call Process
Timer T304	90s	Related to INFO
Timer T305	4s	Related to Disconnect
Timer T308	4s	Related to Release Complete
Timer T309	90s	
Timer T310	20s	Related to ALERT
Timer T313	4s	Related to Connect_ACK and Path Replacement
Timer T306	6s	
Timer T314	2s	
Timer T383	5s	
Timer T389	8s	
Timer T392	1s	
Timer T397	5s	

7.7.5 The SIP proxy

Used to activate some parameters linked to the Proxy (SIP authentication for instance)

SIP initial time-out : This attribute specifies the initial value in milliseconds of the request/reply SIP message retransmission timeout corresponding to T1. Default value 500ms

SIP timer T2 : This attribute specifies the maximum time in milliseconds between two SIP message retransmissions. Default value 4000ms

Dns Timer overflow : Timer used to overflow from DNS 1 to DNS 2

Timer TLS : This attribute is used to define the keep alive for TLS (From R10.0)

Recursive search : This attribute is used to define the behavior of the proxy on reception of a redirection message. (NOT CURRENTLY USED)
- YES: the proxy handles redirection.
- NO: the proxy leaves the caller to handle redirection.

Minimal authentication method : Activation of the Proxy authentication
- SIP none, there is no authentication
- SIP Digest, the authetication is validated

Authentication realm : Corresponds to the authentication SIP domain on the OXE

Only authenticated incoming calls : Activation of the SIP authentication for incoming calls

Framework Period : Indicates the basic time for an observation period before to put the IP address in quarantine (3s by default).

Framework Nb Message By Period : Indicates the maximum number of received messages during the time of the observation periods which may put the IP address in quarantine (25 messages by default).

Framework Quarantine Period : Indicates the periods number before to put the IP address in quarantine (1800s by default)

TCP when long messages : This parameter is used when UDP is used as transport protocol, to allow or not the use of TCP for long messages. This parameter applies to external gateways, SIP extensions, SIP devices and SIP external voice mails.
- True (default value): TCP is used, rather than UDP, when the message size is higher than the maximum size (1300 bytes)
- False: UDP is used, whatever the size of messages.

Retransmission number for INVITE : This Attribute corresponds to the number of INVITE retransmission, from 1 to 6 (From R10.0)

- SIP timers explanation:

Timer	Value	Meaning
Timer 1	500 ms	Round-trip time (RTT) estimate
Timer 2	4000 ms	The maximum retransmit interval for non-INVITE requests and INVITE responses
Timer 4	5000 ms	Maximum duration a message will remain in the network
Timer A	Initially T1	INVITE request retransmit interval, for UDP only
Timer B	64 *T1	INVITE transaction timeout timer
Timer C	> 3 min	Proxy INVITE transaction timeout
Timer D	32s for UDP 0s for TCP	Wait time for response retransmits
Timer E	Initially T1	Non-INVITE request retransmit interval, UDP only
Timer F	64 *T1	Non-INVITE transaction timeout timer
Timer G	Initially T1	INVITE response retransmit interval
Timer H	64 *T1	Wait time for ACK receipt
Timer I	T4 for UDP 0 s for TCP	Wait time for ACK retransmits
Timer J	64* T1 for UDP 0 s for TCP	Wait time for non-INVITE request retransmits
Timer K	T4 for UDP 0 s for TCP	Wait time for response retransmits

7.7.6 SIP Registrar

(idem for R9.1 and R10.x)

Used to manage the registration timers

SIP Min Expiration Date : Minimum lifetime of a record accepted by the Registrar (in secondes). Default value 1800.

SIP Max Expiration Date : Maximum lifetime of a record accepted by the Registrar (in secondes). Default value 86400.

The minimum value must not be under 420 (7 minutes). The REGISTER must not be used for “keep alive” mechanism. 900 (15 minutes) is a minimum acceptable value.

7.7.7 SIP Dictionnary

(idem for R9.1 and R10.x)

Corresponds to the SIP users created on the OXE, this dictionnary is fill up automatically when a SIP user is created, entries on this dictionnary can be created manually if needed (Not used), but the purpose of this object is to be able to modify one entry already created or to add aliases

Directory Number : Corresponds to the directory number of Station, Network number or Vmail number.

Alias No. : Can create different alias for the same directory number

SIP URL Username : User part of the URL. SIP identifies users by their URLs (Universal Resource Locator), composed of a user part and a domain part (user@domain).

SIP URL Domain	: Domain part of the URL. SIP identifies users by their URLs, composed of a user part and a domain part (user@domain). If the domain part is omitted on creation of a set, the domain part of the installation URL is used (SIP/SIPgateway).
SIP URL Type	: Corresponds to the user type (SIP extension or SIP Device).
SIP URL Origin	: Corresponds to the origin node.

7.7.8 SIP Authentication

(idem for R9.1 and R10.x)

Used to modify the password of a entry created automatically (SIP user for instance)

Directory Number	: Directory number of the entry selected (not modifiable)
SIP Authentication	: SIP login associated to the entry (not modifiable)
SIP Passwd	: Enter a new password if needed
Confirm	: Confirmation of the new password entered

7.7.9 Quarantined IP Addresses

(idem for R9.1 and R10.x)

Used to put the IP addresses of the SIP equipments you want to put in quarantined manually, SIP messages from these addresses are dropped silently.

7.7.10 Trusted IP Addresses

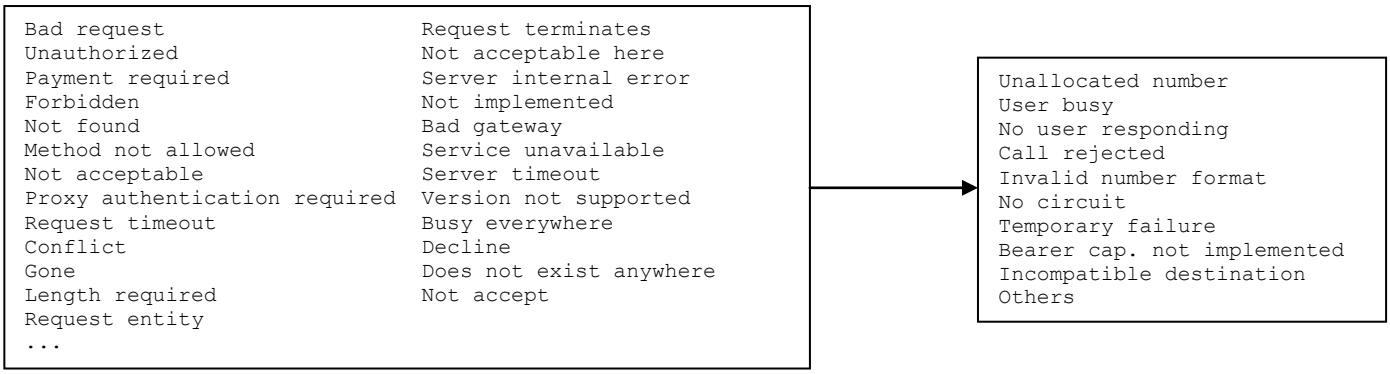
(idem for R9.1 and R10.x)

Used to put the IP addresses of the SIP equipments not affected by the quarantined mechanism. If after management the communication with this SIP equipments is still rejected by the OXE, restart the SIPMOTOR processes.

7.7.11 SIP To CH Error Mapping

(idem for R9.1 and R10.x)

Used to link the error SIP messages to the ISDN Q850 causes, for each error SIP message, you select one Q850 cause
A default configuration is done, without specific needs, no modifications have to be made.



7.7.12 CH To SIP Error Mapping

(idem for R9.1 and R10.x)

Used to link the ISDN Q850 causes to the error SIP messages, for each Q850 cause, you select error SIP message.
A default configuration is done, without specific needs, no modifications have to be done.

```
Unallocated number                          Channel type not implemented
No route to specify transit NW              Req facility not implemented
No route to destination                     Only Rest Digi Info Becap Avail
France Specific                             Option not implemented
Denmark Specific                            Invalid call reference value
Channel unacceptable                        Identified channel does not exist
Call awarded - deliv in estab channel       Susp Call Exists But Call Ident
Reserved MLPP                               Call Identity in use
Normal call clearing                        No call suspended
User busy                                   Call having req call ID cleared
No user responding                          Japan Specific
No answer from user                         Incompatible destination
Call rejected                               Invalid transit network selection
Number changed                              Invalid message
Nonselected user clearing                   Mandatory info element missing
Destination out of order                    Msg type non-exist or not impl
Invalid number format                       Message not compat with call state
Facility rejected                           Info element non-exist or not impl
Response To STATUS INQUIRY                  Invalid info element content
Normal unspecified                          Recovery on timer expiration
No circuit                                  Protocol error
Network out of order                        Interworking
Temporary failure
...
```

→

```
Not found
Gone
Temporarily unavailable
Address Incomplete
Busy here
Not acceptable here
Server internal error
Not implemented
Bad gateway
Service unavailable
Decline
```

7.8.1 SIP Device

The SIP Device is used for voice SIP calls and FAX SIP calls, the SIP Device is considered as an External SIP user, so the features are limited (same as SIP TG)

- SIP Device creation

Directory Number : Corresponds to the directory number of the SIP Device

Set Type : Select “SIP device” for the type of set

URL UserName : The user name corresponds to the SIP Device directory number - autofill

URL Domain : Corresponds to the OXE domaine name (nodename) - autofill

SIP Authentication : The user name corresponds to the SIP Device directory number – autofill

External Gateway Number : Used in case of Open Touch configuration, determine the external Gateway number to reach the OT (From R10.0)

Gateway type : Used in case of Open Touch configuration, determine the gateway type to reach the OT (From R10.0)

- In normal use, only the Directory Number and the set type are managed, the other parameters can be modified only if needed
- The SIP device is linked to the local SIP gateway

The local SIP gateway must be managed and is in service to be able to make and receive calls

With the current Linux OS, OXE has a limitation in handling more than 1000 data equipment if it is connected in the same sub-network. So we need to have a seperate VLAN in between to handle this. OXE CS must be placed under separate subnet and the IP Phones distributed under different other subnets

All unnecessaries subscriptions must be deactivated on SIP Devices when service is not available on OXE, example: Voicemail, notification …

7.8.2 SIP Extension (or SEPLOS)

The SIP Extension is used only for voice calls, is considered as an Internal SIP user, so it is possible to use phone features and facilities from the OXE.

It is not necessary to manage the local SIP gateway if you want to use it, only the proxy has to be (for authentication)

- SIP Extension creation

Directory Number : Corresponds to the directory number of the SIP Extension

Set Type : Select “SIP extension” for the type of set

URL UserName : The user name corresponds to the SIP Extension directory number - autofill

URL Domain : Corresponds to the OXE domain name (nodename) - autofill

SIP Authentication : The user name corresponds to the SIP Extension directory number – autofill

- Other SIP extension parameters

- Under /users/ IP SIP Extension:

Set Type : Type of set displayed (SIP extension or SIP device)

IP Address : IP address of the SIP equipment displayed (information retrevies from the registrar)

- Under /users/ SIP Extension Parameters:

Phone COS (explanation later) : Corresponds to the SIP phone class of service and not the “normal” phone class of service

The SIP extension can be created as a “business” user or “room” user in case of hospitality. One of the difference, it that in case of “business” mode, the SIP extension is multiline (not manageable) and in case of “room” mode , the SIP extension is monoline.

7.9 SIP parameters explanation / under the object SIP Extension:

Used to manage some specific phone features for SIP extension

Display UTF-8 : Used to display UTF-8 name, if the SIP phone is compatible,
- if True, the OXE will send the name in UTF-8 to the SIP Phone
- if False, the OXE will send the “normal” name to the SIP phone

Display call server information : Display information on the set display, for instance if the set is fowarded by using an OXE prefix
- if True, the OXE will send a SIP message MESSAGE
- if False, the OXE will not send this SIP message

- The SIP phone must be compatible with the SIP messages or they will be rejected (405 message).

Example of a message:

```
----------------------utf8-----------------------
MESSAGE sip:31031@172.27.142.64:5060;transport=udp;user=phone SIP/2.0
Supported: replaces,timer,100rel
User-Agent: OmniPCX Enterprise R9.1 i1.605.23
Content-Type: text/plain;charset=UTF-8
To: <sip:31031@myoxe;user=phone>
From: " " <sip:31031@myoxe;user=phone>;tag=40cc45387a17217352a366b1cf047606
Call-ID: ad77e8c268099c4683d97841a78d7ddf@172.27.141.151
CSeq: 1185999967 MESSAGE
Via: SIP/2.0/UDP 172.27.141.151;branch=z9hG4bKb46b58bf397dae02629301df568a1bd7
Content-Length: 26

Immdiate fwd -> 31000
-------------------------------------------------
```

Keep Alive : Used to implement the keep alive mechanism between the OXE to the SIP phone, if the SIP phone is compatible

- if True, the OXE will send an OPTION message to the SIP phone
- if False, the OXE will not send this OPTION message

- The keep alive timer is managed on the IP Quality Of Service COS, assoicated to the IP domain of the SIP Extension user (seen later)

Send NOTIFY instead of MESSAGE : Used to send the synamic state of the SEPLOS SIP message MESSAGE or with a NOTIFY SIP message

(From R10.0)

7.10 SIP parameter explanation / under the object External Voice Mail:

Go under /Applications/ External Voice Mail

Voice Mail Dir.No : Corresponds to the directory number of the External Voice Mail.

Sub Type : - Private (default value): The via header is not used to determine the origin of incoming calls.
- Public: the via header is used to determine the origin of incoming calls when other headers do not match.

URL UserName : Corresponds to the Voice Mail directory number.

URL Domain : Corresponds to the nodename of the OXE.

PCS IP Address : Corresponds to the IP address of the PCS to secure this external SIP Voice Mail.

SIP Authentication : Correponds to the login used for the authentication to the external SIP voice mail

SIP Passwd : Correponds to the password used for the authentication to the external SIP voice mail

Register On Line Number : Directory number used to access the voice mail service in record mode. This number is dialed automatically when the 'Rec.' key is pressed on a set.

Register URL (Username) : User part of the URL used for access to the voice mail service in record mode.

Register URL (Domain) : Domain part of the URL used for access to the voice mail service in record mode.

Register Authentication mode. : Correponds to the login used to control access to the external voice mail service in record

Register Password mode. : Correponds to the password used to control access to the external voice mail service in record

External Gateway Number : Used to manage an entity (SIP Device or External Voice Mail) behind a Proxy. If different from -1, it is used as an ½ Outbound Proxy: outgoing calls are routed to it via its RemoteDomain (Gateway Id) and its Outbound Proxy. Registration (REGISTER) and supervision (OPTIONS) are still configurable.

Subscription on registration : Used if the Subscription is done in the same time than the Registration or in two different messages.

7.11 SIP parameters explanation / under the object System:

Go under /System/Other System Param./SIP Parameters

Packetization times per codec : - If True , as many couple of ptime/maxptime information available for many codecs.
- If False , a single couple of ptime/maxptime information available for many codecs.

Via Header_ Inbound Calls Routing : - If False (default value): The via header is not used to determine the origin of incoming calls.
- If True: the via header is used to determine the origin of incoming calls when other headers do not match with the RemoteDomain of an External Gateway.

Hardwareless for OTBE : NOT CURRENTLY USED (From R10.1)

Local resources : NOT CURRENTLY USED (From R10.1)

Loose Route with RegID : The possibility is offered to accept the call if route only contains a URI with OXE_address without user part.
- If True, INVITE without RegID in route header is re-routed to the destination corresponding to ReqURI domain part.
- If False, INVITE is accepted. (From R10.1)

Reject unidentified proxy calls : As an exceptional procedure for inbound calls, if the origin of the call cannot be determined, either by looking up the SIP dictionary, or through any other procedure (call does not comes from a SIP External Gateway), and if the Source @IP doesn't belong to the trusted @IP list the call is either delivered to the Call Handling on the Main Gateway, or rejected with a 403.Forbidden response.
- If it is set to True, such calls are rejected with a 403.Forbidden response.
- If it is set to False, the call is delivered to the Call Handling on the Main Gateway. (From R10.1)

Transfer : Refer using single step
- If True, new INVITE without Referred-By is provided
- If False, new INVITE with Referred-By is provided (From R10.1)

Go under /System/Other System Param./System Parameters

SRTP TLS offer answer mode : - If True: SRTP according to SDP offer/answer model
- If False: SRTP Oxe centralized SRTP mode
(From R10.0)

TLS signaling possible : - If True: TLS signaling allowed for SIP gateways / TLS signaling and SRTP allowed for SIP sets
- If False: TLS signaling not possible for SIP gateways / TLS signaling and SRTP not possible for SIP
(From R10.0)

Accept Mu and A laws in SIP : From the R9.1, the OXE is using only in G711 the system law for all SIP calls (inbound calls), thanks to this parameter, the OXE is able to accept the G711calls using the other law for inbound calls on external SIP gateways only.

Go under /System/Other System Param./External Signaling Parameters

NPD for External Forward : - If -1: redirection information is sent
- If configured with NPD number used by SIP ISDN Trunk: see the calling name presentation on the set display of called phone in case of forward

Calling Name Presentation : - If False: Calling Number is not sent
- If True: display name to external calls is sent

8. IP DOMAINS, CODECS AND PCS

8.1 IP domains rules

A SIP equipment can belong to an IP domain, according to this configuration, it is able to use some behaviours from its IP domain (see the TC1277 for IP domain configuration and restrictions)

The first thing to know, it is that a SIP equipment doesn't belong to an IP domain if its IP address is not managed, it doesn't belong in the IP domain 0 as well (exept for the SIP extension users acting like IPtouch) in that case if no management is done, the call is everytime an extra domain call with an Alcatel-Lucent equipment.

8.2 System law for PCM codec

The system is accepted only the PCM codec of its law. If the system is using the A law, only PCMA will be accepted and used, PCMU will be rejected.

Exception: for SIP external gateways, if the OXE receives an INVITE with only PCMU, the OXE will accept, but the voice quality is not guaranteed.

The next parameter must be managed:

/System/Other System Param./System Parameters/Accept Mu and A laws in SIP

- False (default): only the system law is accepted
- True: the two laws are accepted

8.3 Codecs on SDP

When a SIP call is done, the OXE manage the SDP according to the next information:

8.3.1 Initial offer : the offer sent in an initial INVITE

The codec list proposed in an initial SDP offer is build according to the algorithm of the outgoing SIP Trunk Group.
The outgoing SIP Trunk Group is the one managed in ARS route or Network/Routing number, **NOT the one managed on the External SIP Gateway**.

This codec list is ordered taking into account calling user extra domain compression law.

Exception : if the caller is a SIP device or a SIP trunk, the codec list is in the same order than the one received from the calling party.

SIP trunk algo must be interpreted as « the best algorithm supported on the trunk » or « the higher bandwidth consumption supported on the trunk» :

- SIP trunk algorithm : default
  - The Trunk Group has low capacity. Only G729/G723 is possible.
- SIP trunk algorithm : G711

- The Trunk Group supports high bandwidth calls and as a consequence low bandwidth calls too. Both G711 and system codec (G729/G723) can be used.

- Initial SDP offer content, general case (calling party is not a SIP device nor a SIP trunk).

Trunk Group compression type	Intra/Extra IP domain algorithm	SDP
Default	With Compression	System algorithm only (G729 for instance)
Default	Without Compression	System algorithm only (G729 for instance)
G711	With Compression	System algorithm (G729 for instance) in first position and PCM (A or MU) in the second position
G711	Without Compression	PCM (A or MU) in first position and system algorithm (G729 for instance) in the second position

8.3.1 nitial answer : the answer to an initial offer on incoming call

Pre-requisite :

- The SIP equipment must at least propose one codec supported by OXE in its offer.
- OXE Trunk Group used for incoming calls (**managed in External SIP Gateway**) must be managed with algo=G711.

OXE always answers with one codec only :

- The one proposed in a by the SIP equipment in case of mono-codec offer.
- The best one in case of multicodec offer, taking into account :
  - SIP equipment list order (calling party prefered codec).
  - Called party extra-domain codec.
- The answer may be send in 18x and/or 200OK depending on « SDP in 18x » management.

- OXE initial SDP answer summary (incoming trunk group algo = G711).

SIP equipment SDP Offer	Intra/Extra IP domain algorithm	Codec use
G729, G711	With Compression	G729
G729, G711	Without Compression	G729
G711, G729	With Compression	G729
G711, G729	Without Compression	G711
G711	With Compression	G711
G711	Without Compression	G711

- For SEPLOS users, the OXE is acting as an IPtouch.

8.4 PCS

The SIP is totally operational on PCS; it is able to secure all types of SIP elements, but the SIP equipment connected must be tested to be sure that it will be able to connect and working on the PCS.

In case of spatial redundancy, the nodename manage on the PCSs must be the same than the one managed on the CPUs.

9. CONTENTS OF A SIP MESSAGES (GENERAL VIEW)

On the SIP messages, we can find different information. According to the type of message, the information can change or can be adapted.

For instance, with an INVITE we can have this:

HEADER

```
INVITE sip:31000@172.27.141.151:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.27.142.64:5060;branch=z9hG4bK3047297329
From: "31031"<sip:31031@172.27.141.151:5060;user=phone>;tag=c0a80101-17193256
To: <sip:31000@172.27.141.151:5060;user=phone>
Call-ID: ebc73a34-c0a80101-0-11@172.27.142.64
CSeq: 1 INVITE
Max-Forwards: 70
Supported: timer, P-Early-Media, replaces
Require: 100rel
Session-Expires: 110
Allow: INVITE,ACK,BYE,CANCEL,OPTIONS,PRACK,SUBSCRIBE,NOTIFY,UPDATE,REFER,REGISTER,INFO
Contact: <sip:31031@172.27.142.64:5060;transport=udp;user=phone>
User-Agent: THOMSON ST2030 hw5 fw2.72 00-1F-9F-16-4F-03
Allow-Events: refer,dialog,message-summary,check-sync,talk,hold
Content-Type: application/sdp
Content-Length: 203
```

BODY

```
v=0
o=MxSIP 4219058434975324735 4219058434975324736 IN IP4 172.27.142.64
s=SIP Call
c=IN IP4 172.27.142.64
t=0 0
m=audio 6000 RTP/AVP 8 0 18 101
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:18 annexb=no
a=fmtp:101 0-15
a=ptime:20
a=mptime:20 20 30 20 -
a=sendrecv
```

Between the Header and the Body, you have everytime an empty line

9.1 The HEADER

The header contains the information to establish a SIP dialog between the UAC and the UAS.

Here the main information given:

- The Request-URI:

```
INVITE sip:31000@172.27.141.151:5060;user=phone SIP/2.0
```

The initial Request-URI of the message SHOULD be set to the value of the URI in the To field, except if the recipient (To field) is forwarded.

- Request-URI: forward destination
- To: forwarded set

- The From:

```
From: "31001"<sip:31031@172.27.141.151:5060;user=phone>;tag=c0a80101-17193256
```

The From header field indicates the logical identity of the initiator of the request.
- The To:

```
To: <sip:31000@172.27.141.151:5060;user=phone>
```

The To header field first and foremost specifies the desired "logical" recipient of the request.

- The Call-ID:

```
Call-ID: ebc73a34-c0a80101-0-11@172.27.142.64
```

The Call-ID header field acts as a unique identifier to group together a series of messages. It MUST be the same for all requests and responses sent by either UA in a dialog.

- The CSeq:

```
CSeq: 1 INVITE
```

A CSeq header field in a request contains a single decimal sequence number and the request method. The CSeq header field serves to order transactions within a dialog, to provide a means to uniquely identify transactions, and to differentiate between new requests and request retransmissions. Two CSeq header fields are considered equal if the sequence number and the request method are identical.

- The Max-Forwards:

```
Max-Forwards: 70
```

The Max-Forwards header field serves to limit the number of hops a request can transit on the way to its destination.

- The Via:

```
Via: SIP/2.0/UDP 172.27.142.64:5060;branch=z9hG4bK3047297329
```

The Via header field indicates the transport used for the transaction and identifies the location where the response is to be sent.

- The Contact:

```
Contact: <sip:31000@172.27.142.64:5060;transport=udp;user=phone>
```

The Contact header field provides a SIP URI that can be used to contact that specific instance of the UA for subsequent requests. Contact header field MUST be present and contain exactly one SIP URI in any request that can result in the establishment of a dialog.

- The Supported and/or Require

```
Supported: timer, P-Early-Media, replaces
```

If the UAC supports (requires) extensions to SIP that can be applied by the server to the response.

- If the UAS receives a supported option tags, it is able to use them if needed.
- If the UAS receives a required option tags, it must use them or reject the request

Other information can appear on header according to the SIP equipment type, to know the meaning of them, check the SIP RFCs

9.2 The BODY

The body contains the message or information used to openan RTP connection (codec, IP address, etc…)

```
v=0
o=MxSIP 4219058434975324735 4219058434975324736 IN IP4 172.27.142.64
s=SIP Call
c=IN IP4 172.27.142.64
t=0 0
m=audio 6000 RTP/AVP 8 0 9 18 101
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:9 G722/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:18 annexb=no
a=fmtp:101 0-15
a=ptime:20
a=mptime:20 20 20 20 20 -
```

SDP session description consists of session-level sections.

Each session-level starts by a letter, corresponding to an information for RTP channel negociation (in voice cases)

In that example, we have the next information given:

- v= : corresponds to SDP version
- o= : corresponds to the originator of the session
    - “MxSIP” = username
    - “4219058434975324735” = sess-id, forms a globally unique identifier for the session
    - “4219058434975324736” = sess-version, is a version number for this session description (increased in case of SDP modification)
    - “IN” = Internet connection type (thru IP network)
    - “IP4” = IP V4 is used for IP addressing
    - “172.27.142.64” = IP address of the SIP equipment (for RTP connection)
- s= : corresponds to the session name
- c= : corresponds to the connection data
    - “IN” = Internet connection type (thru IP network)
    - “IP4” = IP V4 is used for IP addressing
    - “172.27.142.64” = IP address of the SIP equipment (for RTP connection)
- t= : corresponds to the start and stop times for this session (t= <start time> <stop time>)
    - t= 0 0 means that the “timimg” is not used in that case
    - This field is mandatory on SDP

- m= : corresponds to the media description
    - “audio” = media type (audio, video, text,…)
    - “6000” = port number used to sent the media stream
    - “RTP/AVP” = transport protocol, in that case, it is RTP
    - 8 0 9 18 101 = payloads (codecs)
- a= : corresponds to SDP attributes
    - “a=rtpmap:8 PCMA/8000” = codec PCMA available on this SIP equipment
    - “a=rtpmap:0 PCMU/8000” = codec PCMU available on this SIP equipment
    - …
    - “a=rtpmap:101 telephone-event/8000” = payload for DTMF
    - “a=fmtp:18 annexb=no” = no VAD available for this call (annexb)
    - “a=ptime:20” = packet time (framing)
    - “a=mptime:20” = maximum ptime accepted
    - “a=sendrecv” = direction of the call, in that case both directions

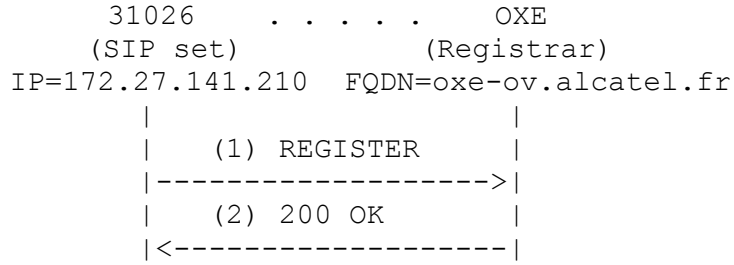
The SDP is generated according to the SIP equipment, each SDP is different for each type of SIP equipment and type of SIP call.

10. EXAMPLES OF COMMON SIP FLOWS

10.1 Registration

In an OmniPCX Enterprise context, the call server (CS) takes the role of the SIP registrar. Registration is necessary to bind a given SIP URL to a physical address. External SIP sets register on the registrar with a SIP **REGISTER** request.
Note that there may be a short delay of several seconds between the time the **REGISTER** message is received and the time the registrar database is updated.

- Without authentication:



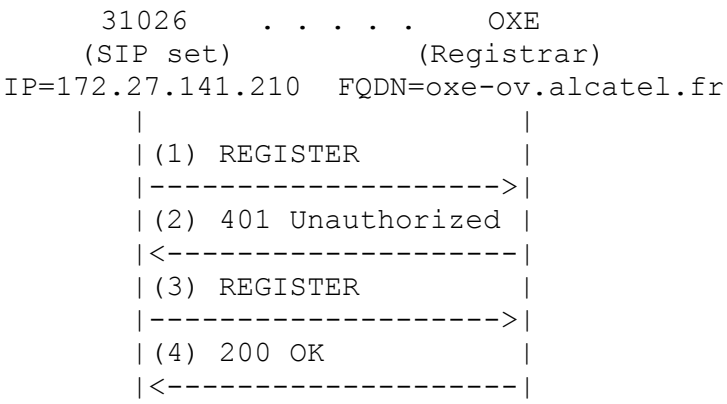

```
----------------------utf8-----------------------
REGISTER sip:oxe-ov.alcatel.fr SIP/2.0
Via: SIP/2.0/UDP 172.27.141.210:22362;branch=z9hG4bK-d87543-826b1a28d80c8c6b-1--d87543-;rport
Max-Forwards: 70
Contact: <sip:31026@172.27.141.210:22362;rinstance=70dae25b3c1e2541>
To: "31026"<sip:31026@oxe-ov.alcatel.fr>
From: "31026"<sip:31026@oxe-ov.alcatel.fr>;tag=e2704074
Call-ID: MDMxYzVkZjBiNWY0NTlmODAwMTk2MTdkNzczZjkwOTM.
CSeq: 1 REGISTER
Expires: 3600
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO
User-Agent: SIP Phone
Content-Length: 0
-------------------------------------------------
```

- The **To** header field contains the address of record (SIP URI) whose registration is to be created. In the example “oxe-ov.alcatel.fr” is the domain (OXE main IP address or FQDN) and 31026 the user name.
- The **Contact** header field contains the physical address (IP address and port) of the record whose registration is to be created. In the example it is 172.27.141.210:22362. Note that if port number would not have been specified it would have been taken as 5060 by default. If any other port number than 5060 is used, it must have to be specified (here 22362).
- The **Expires** field corresponds to the maximum time of registration on the REGISTRAR, the SIP equipment msut send a new REGISTER message to stay on, if not, it will be removed from it.

The registrar answers with a *200 OK* response upon successful registration.

```
----------------------utf8-----------------------
SIP/2.0 200 OK
Contact: <sip:31026@172.27.141.210:22362;rinstance=70dae25b3c1e2541>;expires=3600
To: "31026" <sip:31026@oxe-ov.alcatel.fr>;tag=aacdbedd6291976d33d249b9d1d52820
From: "31026" <sip:31026@oxe-ov.alcatel.fr>;tag=e2704074
Call-ID: MDMxYzVkZjBiNWY0NTlmODAwMTk2MTdkNzczZjkwOTM.
CSeq: 1 REGISTER
Via: SIP/2.0/UDP 172.27.141.210:22362;received=172.27.141.210;branch=z9hG4bK-d87543-e14134135a40db7d-1--d87543-
;rport=22362
Content-Length: 0
```

- With authentication:



The first REGISTER is send without the authenication parameters and the OXE sends a 401 Unauthorized message to ask the SIP equipment for the authentication parameters

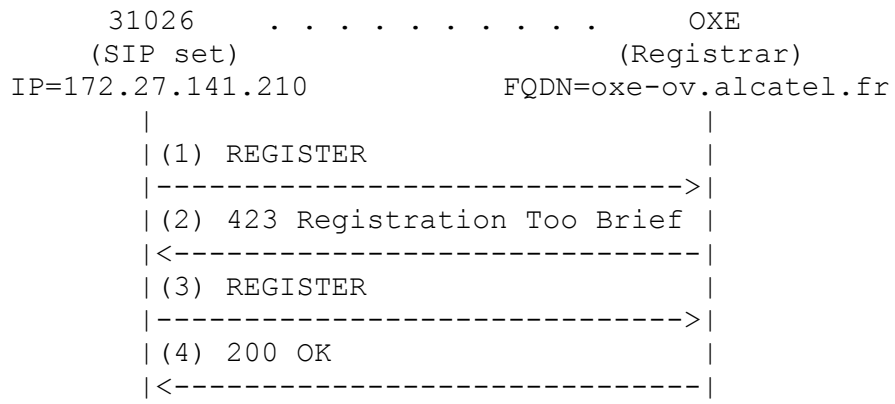
```
----------------------utf8-----------------------
SIP/2.0 401 Unauthorized
WWW-Authenticate: Digest qop="auth",nonce="a4c9e550459f63fd80764dc69609c482",realm="oxe-ov"
To: "31026" <sip:31026@oxe-ov.alcatel.fr>;tag=da389f6e785d72b8910a0f2310d68fcc
From: "31026" <sip:31026@oxe-ov.alcatel.fr>;tag=e2704074
Call-ID: MDMxYzVkZjBiNWY0NTlmODAwMTk2MTdkNzczZjkwOTM.
CSeq: 1 REGISTER
Via: SIP/2.0/UDP 172.27.141.210:22362;received=172.27.141.210;branch=z9hG4bK-d87543-826b1a28d80c8c6b-1--d87543-
;rport=22362
Content-Length: 0
-------------------------------------------------
```

- The **WWW-Authenticate** field corresponds to the OXE information about authentication:The information “**Digest**” corresponds to the challenge type
- The information “**qop**” corresponds to the "quality of protection" values supported by the server. The value "auth" indicates authentication.
- The information “**nonce**” corresponds to control the integrity of the authentication information received by the SIP equipment.
- The information “**realm**” corresponds to the SIP authentication domain, only one can be managed on the OXE.

The Register with the authentication information

```
----------------------utf8-----------------------
REGISTER sip:oxe-ov.alcatel.fr SIP/2.0
Via: SIP/2.0/UDP 172.27.141.210:22362;branch=z9hG4bK-d87543-e14134135a40db7d-1--d87543-;rport
Max-Forwards: 70
Contact: <sip:31026@172.27.141.210:22362;rinstance=70dae25b3c1e2541>
To: "31026"<sip:31026@oxe-ov.alcatel.fr>
From: "31026"<sip:31026@oxe-ov.alcatel.fr>;tag=e2704074
Call-ID: MDMxYzVkZjBiNWY0NTlmODAwMTk2MTdkNzczZjkwOTM.
CSeq: 2 REGISTER
Expires: 3600
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO
User-Agent: SIP Phone
Authorization: Digest username="31026",realm="oxe-ov",nonce="a4c9e550459f63fd80764dc69609c482",uri="sip:oxe-
ov.alcatel.fr",response="dde0d45f751288517
8806dc1b4321b19",cnonce="e53a2b8923348db7",nc=00000001,qop=auth,algorithm=MD5
Content-Length: 0
-------------------------------------------------
```

- When the registration timer is too brief



When the "expires" is too small compares to the OXE one, the OXE returns the message "423 Registration Too Brief", with its timer, in that case, the SIP equipment sends a new REGISTER with the timer received.

```
----------------------utf8-----------------------
REGISTER sip:oxe-ov.alcatel.fr SIP/2.0
Via: SIP/2.0/UDP 172.27.141.210:22362;branch=z9hG4bK-d87543-826b1a28d80c8c6b-1--d87543-;rport
Max-Forwards: 70
Contact: <sip:31026@172.27.141.210:22362;rinstance=70dae25b3c1e2541>
To: "31026"<sip:31026@oxe-ov.alcatel.fr>
From: "31026"<sip:31026@oxe-ov.alcatel.fr>;tag=e2704074
Call-ID: MDMxYzVkZjBiNWY0NTlmODAwMTk2MTdkNzczZjkwOTM.
CSeq: 1 REGISTER
Expires: 60
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO
User-Agent: SIP Phone
Content-Length: 0
-------------------------------------------------
```

- The "**Expires**" value is equal to 60 in that case, and the minimum value managed on the OXE is 1800

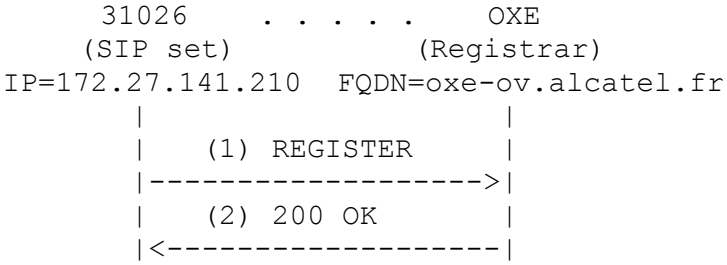
```
----------------------utf8-----------------------
SIP/2.0 423 Registration Too Brief
Min-Expires: 1800
To: "31026"<sip:31026@oxe-ov.alcatel.fr>;tag=85d8c7828811c12691305052d6ef7f9a
From: "31026"<sip:31026@oxe-ov.alcatel.fr>;tag=e2704074
Call-ID: MDMxYzVkZjBiNWY0NTlmODAwMTk2MTdkNzczZjkwOTM.
CSeq: 1 REGISTER
Via: SIP/2.0/UDP 172.27.141.210:22362;branch=z9hG4bK-d87543-826b1a28d80c8c6b-1--d87543-;rport
Content-Length: 0
-------------------------------------------------
```

- The information "**Min-Expires**" correponds to the minimun registration timer value of the OXE (manage on the REGISTRAR object)

```
----------------------utf8-----------------------
REGISTER sip:oxe-ov.alcatel.fr SIP/2.0
Via: SIP/2.0/UDP 172.27.141.210:22362;branch=z9hG4bK-d87543-826b1a28d80c8c6b-1--d87543-;rport
Max-Forwards: 70
Contact: <sip:31026@172.27.141.210:22362;rinstance=70dae25b3c1e2541>
To: "31026"<sip:31026@oxe-ov.alcatel.fr>
From: "31026"<sip:31026@oxe-ov.alcatel.fr>;tag=e2704074
Call-ID: MDMxYzVkZjBiNWY0NTlmODAwMTk2MTdkNzczZjkwOTM.
CSeq: 1 REGISTER
Expires: 1800
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO
User-Agent: SIP Phone
Content-Length: 0
-------------------------------------------------
```

- The new REGISTER received on the OXE has the value 1800 (the one from the message 423)

10.2 De-registration



When a SIP equipment is stopped, before it has to send a REGISTER message to be removed from the OXE REGISTRAR, for this, it has to send a REGISTER with an “Expires = 0”

```
----------------------utf8-----------------------
REGISTER sip:oxe-ov.alcatel.fr SIP/2.0
Via: SIP/2.0/UDP 172.27.141.210:22362;branch=z9hG4bK-d87543-826b1a28d80c8c6b-1--d87543-;rport
Max-Forwards: 70
Contact: <sip:31026@172.27.141.210:22362;rinstance=70dae25b3c1e2541>
To: "31026"<sip:31026@oxe-ov.alcatel.fr>
From: "31026"<sip:31026@oxe-ov.alcatel.fr>;tag=e2704074
Call-ID: MDMxYzVkZjBiNWY0NTlmODAwMTk2MTdkNzczZjkwOTM.
CSeq: 1 REGISTER
Expires: 0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO
User-Agent: SIP Phone
Content-Length: 0
-------------------------------------------------
```

- On the REGISTER, we have the “**Expires = 0**” and the “**Contact**”, this contact is used by the REGISTRAR to know which physical IP address to remove for this URI (in case of forking).
- If the “**Contact**” is received with a “*****”, the REGISTRAR must removed all the “**Contact**” associated.

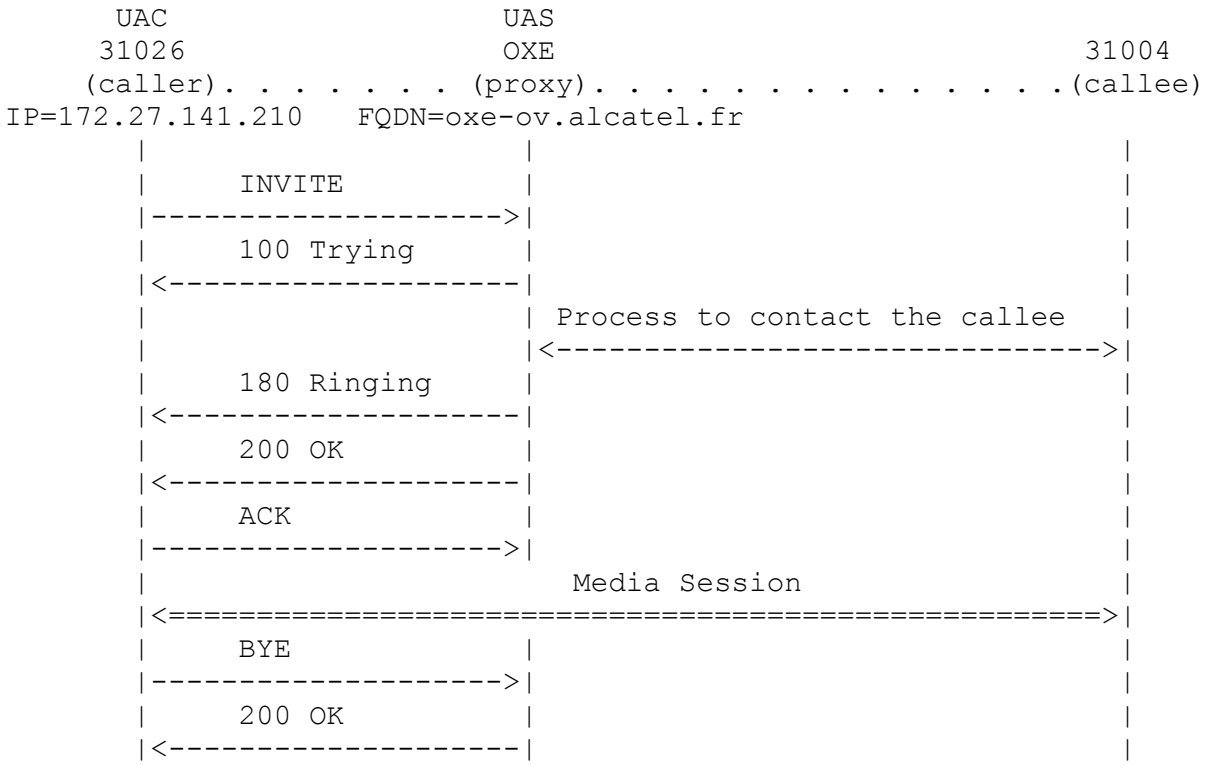
In case of duplication, when the Main CPU receives a REGISTER, the SIPMOTOR sends this REGISTER to the Stand-BY CPU with the next message:

```
----------------------utf8-----------------------
REGISTER sip:oxe-ov.alcatel.fr SIP/2.0
Via: SIP/2.0/UDP 172.27.141.210:22362;received=172.27.141.210;branch=z9hG4bK-d87543-e14134135a40db7d-1--d87543-
;rport=22362
Max-Forwards: 70
Contact: <sip:31026@172.27.141.210:22362;rinstance=70dae25b3c1e2541>;expires=3600
To: "31026" <sip:31026@oxe-ov.alcatel.fr>
From: "31026" <sip:31026@oxe-ov.alcatel.fr>;tag=e2704074
Call-ID: MDMxYzVkZjBiNWY0NTlmODAwMTk2MTdkNzczZjkwOTM.
CSeq: 2 REGISTER
Expires: 3600
Allow: INVITE
Allow: ACK
Allow: CANCEL
Allow: OPTIONS
Allow: BYE
Allow: REFER
Allow: NOTIFY
Allow: MESSAGE
Allow: SUBSCRIBE
Allow: INFO
Content-Length: 0
User-Agent: Alcatel-main Registrar

-------------------------------------------------
```

10.3 Simple call establishement

The following diagram shows the messages sent from a SIP equipment to an OXE user (Not a SIP one)



1) The SIP equipment sends an INVITE to the OXE

```
Mon Jun 25 11:10:17 2012 RECEIVE MESSAGE FROM NETWORK (172.27.141.210:63016 [UDP])
----------------------utf8-----------------------
INVITE sip:31004@oxe-ov.alcatel.fr SIP/2.0
Via: SIP/2.0/UDP 172.27.141.210:63016;branch=z9hG4bK-d87543-4c3f8f26d532b437-1--d87543-;rport
Max-Forwards: 70
Contact: <sip:31026@172.27.141.210:63016>
To: "31004"<sip:31004@oxe-ov.alcatel.fr>
From: "31026"<sip:31026@oxe-ov.alcatel.fr>;tag=e9708b0f
Call-ID: MzI0MjQ4MmQ5NjMzZTVmZTlmYTE5NTVhMGNiZWI0ODQ.
CSeq: 1 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO
Content-Type: application/sdp
User-Agent: SIP Phone
Content-Length: 417

v=0
o=- 6 2 IN IP4 172.27.141.210
s= SIP Phone
c=IN IP4 172.27.141.210
t=0 0
m=audio 52694 RTP/AVP 18 101
a=fmtp:18 annexb=no
a=fmtp:101 0-15
a=rtpmap:101 telephone-event/8000
a=sendrecv
------------------------------------------------
```

- The INVITE can contain SDP or not. If there is no SDP, the ACK (after the 200ok) sent must contain the SDP information

2) The SIP equipment receives a provisional answer from the OXE (100 Trying)

```
Mon Jun 25 11:10:17 2012 SEND MESSAGE TO NETWORK (172.27.141.210:63016 [UDP]) (BUFF LEN = 338)
----------------------utf8-----------------------
SIP/2.0 100 Trying
To: "31004" <sip:31004@oxe-ov.alcatel.fr>
From: "31026" <sip:31026@oxe-ov.alcatel.fr>;tag=e9708b0f
Call-ID: MzI0MjQ4MmQ5NjMzZTVmZTlmYTE5NTVhMGNiZWI0ODQ.
CSeq: 1 INVITE
Via: SIP/2.0/UDP 172.27.141.210:63016;received=172.27.141.210;branch=z9hG4bK-d87543-4c3f8f26d532b437-1--d87543-
;rport=63016
Content-Length: 0
-------------------------------------------------
```

- The 100 Trying is a provisional message sent by the OXE, this message is generated by the SIPMOTOR directly, it can be concidered as an automatic answer of an INVITE to avoid retransmission from UAC.

3) The SIP equipment receives a provisional answer from the OXE (180 Ringing or 183 Session Progress)

```
Mon Jun 25 11:10:18 2012 SEND MESSAGE TO NETWORK (172.27.141.210:63016 [UDP]) (BUFF LEN = 815)
----------------------utf8-----------------------
SIP/2.0 180 Ringing
Allow: INVITE, ACK, CANCEL, BYE, PRACK, NOTIFY, REFER, SUBSCRIBE, OPTIONS, UPDATE
Contact: sip:oxe-ov.alcatel.fr
User-Agent: OmniPCX Enterprise R10.0 j1.410.45
Content-Type: application/sdp
To: "31004" <sip:31004@oxe-ov.alcatel.fr>;tag=bb28096d41c595340f577a538bf30d54
From: "31026" <sip:31026@oxe-ov.alcatel.fr>;tag=e9708b0f
Call-ID: MzI0MjQ4MmQ5NjMzZTVmZTlmYTE5NTVhMGNiZWI0ODQ.
CSeq: 1 INVITE
Via: SIP/2.0/UDP 172.27.141.210:63016;received=172.27.141.210;branch=z9hG4bK-d87543-88163a3aa534591a-1--d87543-
;rport=63016
Content-Length: 0
-------------------------------------------------
```

- The 180 Ringing (or 183 Progress Session) is a provisional message sent by the OXE, this message is used to inform the caller, that the remote party is ringing. This message can contain SDP to provide the Ring back tone RBT), if no SDP, the RBT must be played localy on the system initator of the call.

4) The SIP equipment receives a 200ok answer from the OXE

```
Mon Jun 25 11:10:19 2012 SEND MESSAGE TO NETWORK (172.27.141.210:63016 [UDP]) (BUFF LEN = 972)
----------------------utf8-----------------------
SIP/2.0 200 OK
Allow: INVITE, ACK, CANCEL, BYE, PRACK, NOTIFY, REFER, SUBSCRIBE, OPTIONS, UPDATE
Contact: sip:oxe-ov.alcatel.fr
Supported: replaces,timer,path,100rel
User-Agent: OmniPCX Enterprise R10.0 j1.410.45
Session-Expires: 1800;refresher=uas
P-Asserted-Identity: "IPtouch 172.27.142.64" <sip:31004@oxe-ov.alcatel.fr;user=phone>
Content-Type: application/sdp
To: "31004" <sip:31004@oxe-ov.alcatel.fr>;tag=bb28096d41c595340f577a538bf30d54
From: "31026" <sip:31026@oxe-ov.alcatel.fr>;tag=e9708b0f
Call-ID: MzI0MjQ4MmQ5NjMzZTVmZTlmYTE5NTVhMGNiZWI0ODQ.
CSeq: 1 INVITE
Via: SIP/2.0/UDP 172.27.141.210:63016;received=172.27.141.210;branch=z9hG4bK-d87543-88163a3aa534591a-1--d87543-
;rport=63016
Content-Length: 242

v=0
o=OXE 1340615417 1340615418 IN IP4 172.27.141.151
s=abs
c=IN IP4 172.27.142.64
t=0 0
m=audio 32514 RTP/AVP 18 101
a=sendrecv
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=ptime:20
a=maxptime:40
a=rtpmap:101 telephone-event/8000
------------------------------------------------
```

- The 200ok is used to open the SIP dialog (in that case), when the called party hang up, the OXE sends this 200ok with a SDP to provide the RTP information for connection.

6) The SIP equipment sends a ACK to the OXE

```
Mon Jun 25 11:10:19 2012 RECEIVE MESSAGE FROM NETWORK (172.27.141.210:63016 [UDP])
----------------------utf8-----------------------
ACK sip:oxe-ov.alcatel.fr SIP/2.0
Via: SIP/2.0/UDP 172.27.141.210:63016;branch=z9hG4bK-d87543-342bae0b06436266-1--d87543-;rport
Max-Forwards: 70
Contact: <sip:31026@172.27.141.210:63016>
To: "31004"<sip:31004@oxe-ov.alcatel.fr>;tag=bb28096d41c595340f577a538bf30d54
From: "31026"<sip:31026@oxe-ov.alcatel.fr>;tag=e9708b0f
Call-ID: MzI0MjQ4MmQ5NjMzZTVmZTlmYTE5NTVhMGNiZWI0ODQ.
CSeq: 1 ACK
User-Agent: SIP Phone
Content-Length: 0
------------------------------------------------
```

- The ACK is used to confirm the dialog. The ACK must contain a SDP if there is no SDP on the INVITE

7) The SIP equipment can send or receive a BYE, when the call is stopped

```
1340615420 -> Mon Jun 25 11:10:20 2012 SEND MESSAGE TO NETWORK (172.27.141.210:63016 [UDP]) (BUFF LEN = 454)
----------------------utf8-----------------------
BYE sip:31026@172.27.141.210:63016 SIP/2.0
Supported: replaces,timer,path,100rel
User-Agent: OmniPCX Enterprise R10.0 j1.410.45
To: sip:31026@oxe-ov.alcatel.fr;tag=e9708b0f
From: "31004" <sip:31004@oxe-ov.alcatel.fr>;tag=bb28096d41c595340f577a538bf30d54
Call-ID: MzI0MjQ4MmQ5NjMzZTVmZTlmYTE5NTVhMGNiZWI0ODQ.
CSeq: 394681697 BYE
Via: SIP/2.0/UDP 172.27.141.151;branch=z9hG4bK07f995532987067a46049349aeaf0592
Max-Forwards: 70
Content-Length: 0
------------------------------------------------
```

- The BYE is used to stop the dialog

8) The SIP equipment can send or receive a 200ok, to confirm the BYE

```
Mon Jun 25 11:10:20 2012 RECEIVE MESSAGE FROM NETWORK (172.27.141.210:63016 [UDP])
----------------------utf8-----------------------
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.27.141.151;branch=z9hG4bK07f995532987067a46049349aeaf0592
Contact: <sip:31026@172.27.141.210:63016>
To: <sip:31026@oxe-ov.alcatel.fr>;tag=e9708b0f
From: "31004"<sip:31004@oxe-ov.alcatel.fr>;tag=bb28096d41c595340f577a538bf30d54
Call-ID: MzI0MjQ4MmQ5NjMzZTVmZTlmYTE5NTVhMGNiZWI0ODQ.
CSeq: 394681697 BYE
User-Agent: SIP Phone
Content-Length: 0
------------------------------------------------
```

11. TROUBLESHOOTING

This section provides step-by-step instructions and troubleshooting actions when you run into trouble.

When a SIP issue is present on the OXE, it is necessary to find the cause of this trouble. To do this, it necessary to make some investigations to find it.

Regarding the issue, different ways of investigations are possible.

- SIP call not possible
- Voice problem
- Fax transmission problem
- DTMF issue
- SIPMOTOR “crash”
- ...

Before to start, here some explainations about the SIPMOTOR functionning and the traces in case of SIP calls.

11.1 SIPMOTOR processes

The first step is to check if all the SIPMOTOR processes are running well on the OXE.

For this, you can use the command “ps -edf | grep sip”, can use the command “pidof sipmotor” to get all the pid numbers.

In R9.1:

```
(1)OXE> ps -edf | grep sip
root      2033   822  0 Feb22 ?        00:00:00 /DHS3bin/servers/sipmotor
root      2139  2033  0 Feb22 ?        00:00:07 /DHS3bin/servers/sipmotor
root      2140  2033  0 Feb22 ?        00:00:00 /DHS3bin/servers/sipmotor
root      2141  2033  0 Feb22 ?        00:00:00 /DHS3bin/servers/sipmotor
root      2142  2033  0 Feb22 ?        00:00:00 /DHS3bin/servers/sipmotor
```

In R10.0

```
(1)OXE> ps -edf | grep sip
root      2202   801  0  2011 ?        00:00:00 [#sipmotor]
root      2203  2202  0  2011 ?        00:00:00 [sipmotor_tcl]
root      2204  2202  0  2011 ?        00:00:00 [sipmotor]
root      2205  2202  0  2011 ?        00:00:00 [sipmotor_dump]
root      2206  2202  0  2011 ?        00:00:00 [sipmotor_presen]
```

In normal functionning, the system displays the sipmotor processes, we have 5 processes and the owner of the processes is root (before the R9.1, the owner was mtcl). According to the OXE release/version, the number of processe can be different.

- If the command gives you this result:

```
(1)OXE> ps -edf | grep sip
root      2033   822  0 Feb22 ?        00:00:00 /DHS3bin/servers/sipmotor
root      2139  2033  0 Feb22 ?        00:00:07 /DHS3bin/servers/sipmotor
mtcl     11942 10204  0 09:40 pts/0    00:00:00 grep sip
```

In that case, you don't have the good number of processes, you can make a double bascul or a reboot the CPU must be performed (shutdown -r 0).

- If you run the command, and you have the next result:

```
(1)OXE> ps -edf | grep sip
root      2033   822  0 Feb22 ?        00:00:00 [#sipmotor <defunct>]
mtcl     12400 10204  0 09:53 pts/0    00:00:00 grep sip
```

In that case, the SIPMOTOR processes have been restarded (automatically or manually), but the configuration of the SIP is not well done, so the configuration must be checked:

- The configuration of the SIP trunk group, used on the local SIP gateway (node number, etc…).
- The configuration of the local SIP gateway is well done (good SIP trunk group used, etc…).

After modifications, the OXE must be rebooted (shutdown -r 0).

11.2 SIPMOTOR memory used

When a problem is present on SIP, it is important to check the use of memory for the SIPMOTOR.
For this, run the command top -p "PID of the SIPMOTOR".

```
10:35am  up 3 days, 19:49,  1 user,  load average: 9.00, 9.00, 9.00
1 processes: 1 sleeping, 0 running, 0 zombie, 0 stopped
CPU states:  0.0% user,  0.0% system,  0.0% nice, 100.0% idle
Mem:   901304K av,  275124K used,  626180K free,       0K shrd,    4752K buff
Swap: 1052216K av,    2180K used, 1050036K free                  177596K cached

  PID USER     CLS  PRI  NI  SIZE  RSS SHARE STAT %CPU %MEM   TIME COMMAND
27956 root     FIFO  99 -12  3996 3996  3616 S.<   0.0  0.4   0:00 #sipmotor
```

The information to check are the "%CPU" and "%MEM":

- If they are increasing when the traffic is more and more higher and decreasing when the traffic is going down, it seems that there is no issue present about memory leak.
- If they are increasing continously, even if there is no traffic, in that case a problem is present, and a SR must be opened for analyse.

When memory leak is present, swap partition incidents are also generated. If the next message is present, check with the command top to see if the SIPMOTOR is using to much memory.

```
20/03/12 15:15:24 000002M|---/--/-/---|=2:2071=Swap partition 24 per cent full
```

11.3 Check the SYSTEM and SIPMOTOR backtraces/alarms

11.3.1 Backtraces

- "excvisu"

The excvisu can be used to see if system backtraces have been generated by the OXE.
To know if the backtrace is about SIP, checks the next information:

- "SIPM", it means that the backtrace is on the SIPMOTOR itself.

```
==============================
There is a new exception. Its address is : 0XBFFFF118 in SIPM. Monitel time : 024283. Date : Tue Apr  6 10:46:40
2010
Application-exception no 11 in SIPM, PC=0xbffff118:3221221656 --> _end
 * SIPM Backtrace: 0x081631c8:135672264 --> CResponse::create
 * SIPM Backtrace: 0x08185ce0:135814368 --> CTransProceedingState::createResponse
 * SIPM Backtrace: 0x08152c09:135605257 --> CTransaction::createResponse
 * SIPM Backtrace: 0x0814d3bf:135582655 --> CDialog::createResponse
 * SIPM Backtrace: 0x0816ab94:135703444 --> CCall::makeGenericResponse
 * SIPM Backtrace: 0x080e8f8b:135171979 --> CMotorCall::makeResponse
 * SIPM Backtrace: 0x080e642e:135160878 --> CMotorCall::emitServerFailureMessage
```

- Backtrace for SIP Extension, the subtype information contains "SIP_EXTENSION"

```
==============================
There is a new exception. Its address is : 0X092EEAA3. Monitel time : 1961696. D
ate : Thu Feb 21 18:45:46 2008
Applicative-Error-Backtrace, thread 1371, PC=0x092eeaa3:154069667, eqt=1380, ser
v=0 --> Kb_Interro
Eqt type=POS_NUM, cr=4, cpl=0, der_us=0, term=12, subtype = SIP_EXTENSION
 * Backtrace: 0x082f9c8e:137337998 EBP 0x01856e94 --> egzis_li
 * Backtrace: 0x08ae2b6b:145632107 EBP 0x01856ea8 --> testprio
 * Backtrace: 0x09263051:153497681 EBP 0x01856ec0 --> real_main
GS=0000055b   FS=00003039   ES=00000000   DS=0000002b
EDI=09823194  ESI=00000000  EBP=01856e94 ESP0=00000000
EBX=538c4e97  EDX=538ca920  ECX=210b3100  EAX=00000019
EIP=092eeaa3   CS=00000023 ESP3=01856e78 EFLAGS=00000246
Continuing at previous PC=0x092eeaa3:154069667
```

- If the address start by "cr=19" the backtrace can be linked to the SIP Trunk Group, the cr=19 corresponds to the virtual shelf for the IP-Link, so the Backtrace could be for another feature using the IP-Link, use the command "trkvisu" to see if the position ("cr" + "cpl" + "term") corresponds to the SIP Trunk Group.

```
==============================
There is a new exception. Its address is : 0X093C1E26. Monitel time : 250434. Date : Tue Mar 20 09:18:48 2012
Application-exception no 5, thd 1176, PC=0x093c1e26:154934822, eqt=13517, serv=0 --> __CHECK__
Eqt type=JONCT, cr=19, cpl=0, der_us=0, term=2
 * Backtrace: 0x08333369:137573225 EBP 0x01826db8 --> nuphmult
 * Backtrace: 0x08990328:144245544 EBP 0x01826ddc --> process_ccbs_exec_poss
 * Backtrace: 0x08999135:144281909 EBP 0x01826e30 --> analyse_facilite_abc
 * Backtrace: 0x08999a05:144284165 EBP 0x01826e3c --> analyse_facilite
 * Backtrace: 0x087fc81d:142592029 EBP 0x01826e4c --> arr_ipns
 * Backtrace: 0x08836851:142829649 EBP 0x01826e7c --> sui_arr_q931
 * Backtrace: 0x08836b09:142830345 EBP 0x01826eac --> arr_q931
```

For all backtraces about SIP, a SR must be opened for R&D analyses.

- “sipmotor.crash”

Under /tmpd, there is a file called “sipmotor.crash” containing the SIPMOTOR “crash” information (file includes on the Infocollect).

```
(1)OXE> more /tmpd/sipmotor.crash
sipmotor.crash generated at Tue Oct 19 09:15:42 2010
1287472542 -> [CMotorCallManager::insertCallwithEqt] CMotorCall 1911 inserted.4NzQyZjY2NTI2ZT
1287472542 -> [quoteString] => "31017"onse]Trying to find the right dialogte = Terminated, cu
1287472542 -> 1186[CMotorCall::inviteBuildFromAssertedId] no P_Asserted_Identity c33435cb1ed7
1287472542 -> 1186[CMotorCall::setFilterUsedMode] To be traced = 0undterminated reason : None
```

If the sipmotor.crash file increase after SIP calls, to see which calls are causing this, make SIPMOTOR traces, all the information present in this file, are taken from the SIPMOTOR, and seen on the traces.

11.3.2 Alarms

On the OXE, some SIP incidents can be generated, next, the explanation of each one.

- 5800: “X” SIP trunk group put into service.

This incident is used to informed you that the SIP trunk group “X” is put in service.

- 5801: “X” SIP trunk group put out of service.

This incident is used to informed you that the SIP trunk group “X” is put out of service, if the trunk group is put out of service automaticaly by the OXE (without human action) open a SR for analyses.

- 5812: SIP external gateway “Y” is in service.

This incident is used to informed you that the SIP gateway “Y” is in service.

- 5813: SIP external gateway “Y” is out of service.

This incident is used to informed you that the SIP external gateway “Y” is put out of service, if the external SIP gateway is put out of service automaticaly by the OXE (without human action) open a SR for analyses.

The state of the SIP Trunk Group and the external SIP gateway are linked:

- If the SIP Trunk Group associated to the SIP external gateway is out of service, the SIP external gateway is out of service too.
- If the external SIP gateway is out of service, the SIP trunk group associated is out of service also, except if this SIP Trunk Group is associated to another external SIP gateway, and this one is in service.
- If all the external SIP gateway associated with one SIP Trunk Group are out of service, the SIP Trunk group will be out of service.

- 5814: Critical failure in SIP component.
- 5815: Major failure in SIP component.
- 5816: Minor failure in SIP component.

These 3 incidents give an information about a problem during SIP exchanges (Registrations, Calls, etc...).

To get more information about thes incidents, go under /tmpd/ and open the sipalarm files.

```
(1)cpua_ov> ll sipal*
-rw-rw-rw-    1 root     tel         15658 Feb 23 09:54 sipalarm.log
-rw-rw-rw-    1 root     root        20456 Nov 10 11:48 sipalarm1.log
-rw-rw-rw-    1 root     tel         20529 Nov 10 12:30 sipalarm2.log
-rw-rw-rw-    1 root     tel         20529 Nov 10 13:28 sipalarm3.log
-rw-rw-rw-    1 root     root        20553 Nov  2 09:17 sipalarm4.log
-rw-rw-rw-    1 root     root        20553 Oct 30 15:29 sipalarm5.log
-rw-rw-rw-    1 root     root        20553 Oct 30 23:47 sipalarm6.log
-rw-rw-rw-    1 root     root        20553 Oct 31 07:16 sipalarm7.log
-rw-rw-rw-    1 root     root        20553 Oct 31 15:38 sipalarm8.log
-rw-rw-rw-    1 root     root        20553 Oct 31 23:59 sipalarm9.log
```

The file sipalarm.log corresponds to the current one.

To make the link between the incident and an entrie in the sipalarm file, check the date and hour of the incident generation with incvisu:

```
01/14/11 15:46:02 000001M|---/--/-/---|=2:5816=Minor failure in SIP component
```

After check on the sipalam file the entry in this time:

```
> 01/14/11 - 15:46:02   Minor alarm
        [receiveInviteEvent] Call:  eqt: 1674 INITIAL_STATE failed to emit an Invite message.
```

In that case, the SIPMOTOR was not able to send an INVITE (lake of licenses for instance).

When the incidents 5814, 5815 and 5816 are generated, and if you constat some problems on the OXE, a SR can be open with the information from the command incvisu and the sipalarm files (or send the Infocollect).

11.4 SIP traces

The OXE has different levels of traces to get information from the different elements (SIPMOTOR, Call handling, IP).
The traces can be run on the Main CPU and on the Stand-By CPU.

11.4.1 SIPMOTOR traces

The SIPMOTOR traces are used to make traces at the sipmotor level, for this, the command “motortrace” can be used to set the level of trace you need.

```
motortrace (v5.2.0) verbosity = 00000000
Correct usage is:
    motortrace trace-level  To set the current trace level.
    motortrace +T_TRACE     To add a single level to the current trace.
    motortrace -T_TRACE     To remove a single level to the current trace.

    T_MOTOR, T_SIP, T_PKT_IN, T_PKT_OUT, T_IPC_IN, T_IPC_OUT, T_INTERNAL_DESTR,
    T_LOG, T_DEBUG, T_FW, T_DB, T_US, T_MOTOR_TEST, T_TRANSPORT, T_ADNS

trace-level :
    0 : No trace (only Alarm)
    1 : Basic trace  (T_PKT_IN|T_PKT_OUT|T_IPC_IN|T_IPC_OUT)
    2 : Medium trace (T_MOTOR|T_SIP|T_PKT_IN|T_PKT_OUT|T_IPC_IN|T_IPC_OUT)
    3 : All traces

    4 : Medium trace dupli (T_MOTOR_TEST|T_SIP|T_PKT_IN|T_PKT_OUT|T_IPC_IN|T_IPC_OUT)
    5 : All traces + dupli
    6 : All traces + T_TRANSPORT + T_ADNS
    7 : All traces + T_INTERNAL_STRUCT
    8 : Medium trace options (T_MOTOR|T_SIP|T_PKT_IN|T_PKT_OUT|T_IPC_IN|T_IPC_OUT|T_OPTIONS_OPTIM)
    9 : All traces + options
    c : Configuration
Traces will be directed to the window, where traced is executed (TL).
Current level of trace is:
sipmotor trace-level 0 (No trace).
```

According to the level of traces, the information given are different.

- If you select 0, that means you have no SIP traces, only the alarms are displayed
- If you select 1, we have only the SIP messages and the information given by the Call Handling
- If you select 2, we have more information given, compares to the level 1, we can see the SIPMOTOR checking the external SIP gateway associated to the INVITE received (for instance)
- If you select 3, we have all the SIP traces, this level is the most used.
- If you select 4, we have the level 2 traces + the duplication information (SIP exchanges between the Main and the Stand-By CPUs)
- If you select 5, we have the level 3 traces + the duplication information (SIP exchanges between the Main and the Stand-By CPUs)
- If you select 6, we have all the traces + the transport trace (network) + the DNS information
- If you select 7, we have all the traces + the internal structure of SIP in SIPMOTOR (From R10.0)
- If you select 8, we have the level 2 traces + options (From R10.0)
- If you select 9, we have all the traces + all options (From R10.0)

When you increase the level for the traces, you increase also the size of the traces.

The command “traced” is used to output the traces, some options are possibles:

- If you use the “traced &”, the trace is running in background.
- If you use the “traced >/tmpd/name_of_the_file.log” to put the trace in a file.
- If you use the “traced -1 /tmpd/traced -s 10000000 -f 50 -d”, to make rotating trace.

```
(1)OXE> motortrace 3
motortrace (v5.2.0) verbosity = 0037b524
sipmotor trace-level set 3 (All traces).
(1)OXE> traced
** UNIX-trace-daemon started ... (static user group No 1) **

traced started ...
```

Make a “CTRL + C” to stop the trace or “killall traced” when the trace is running in background.

The level of traces must be put back motortrace 0 after traces taken to avoid memory leak.

- The option “c” can be used to display all the SIP configuration (local)

```
(1)OXE> motortrace c
motortrace (v5.2.0) verbosity = 0037b524
sipmotor trace-level set c (data dump).

Proxy parameters.
=================
    sip stack version      4.0.006.022
    initial_timeout        500
    timer_t2               4000
    recursion              0
    min_auth_method        0 NONE=0 DIGEST=2
    auth_realm             cpua
    sipDnsTimerPrimSecond  5000
    onlyAuthIncomingCalls  1
    quarantine and trusted addresses:
    nb_msg_by_period             25
    period                       3
    framework_quarantine_period  1800

Gateway parameters.
===================
    url_install            172.27.141.151
    url_gw
    url_hostname           oxe-ov
    num_ss_reseau          1
    num_faisc              10
    proxy_address           not used
    DNS_localDomName       alcatel.fr
    DNS_type               0 dnsa=0, dnssrv=1
    DNS_primaire           Unused
    DNS_secondaire         Unused
    prack_required         0
    out_proxy              0 AUCUN=0 INTEGRE=1 EXTERNE=2
    proxy_port             5060
    proxy_transport        1 TCP=0 UDP=1
    sipSubsMinDuration     1800
    sipSubsMaxDuration     86400
    sipSessionTimer        1800
    sipMinSessionTimer     900
    SessionTimerMethod     1 re-invite=0, update=1
    sipCac                 0
    SDP_in_180             0
    sip_info_enable        0
    payload                97
    seplos                 1

...
```

11.4.2 Call Handling traces

Call Handling traces can be provided in case of issue, because there is a permanent link between the Call Handling and the SIPMOTOR, so the call can be rejected by the Call Handling and not from the SIPMOTOR.

The SIPMOTOR traces and the Call Handling traces must be done simultaneously.

Here, the basic Call Handling trace done on the OXE.

```
(1)OXE> tuner km
(1)OXE> tuner all=off
(1)OXE> tuner clear-traces
(1)OXE> trc i
+--------+-------+--------+--------+---------+---------+----------+------+
| filter | desti | src_id | cr_nbr | cpl_nbr | us_term | term_nbr | type |
+--------+-------+--------+--------+---------+---------+----------+------+
|    0   |       |        |        |         |         |          |      |
|    1   |       |        |        |         |         |          |      |
|    2   |       |        |        |         |         |          |      |
|    3   |       |        |        |         |         |          |      |
|    4   |       |        |        |         |         |          |      |
|    5   |       |        |        |         |         |          |      |
|    6   |       |        |        |         |         |          |      |
|    7   |       |        |        |         |         |          |      |
+--------+-------+--------+--------+---------+---------+----------+------+
(1)OXE> tuner +cpu +cpl +at +time hybrid=on
(1)OXE> actdbg all=off

Thu Feb 24 10:41:42 CET 2011

(1)OXE> actdbg sip=on

Thu Feb 24 10:41:52 CET 2011

(1)OXE> mtracer -a
Traces Analyser activated

mtracer started ...
(858432:000001) MTRACER host (172.27.141.149, OXE), version: R9.1-i1.605-23-fr-c0
```

After, according to the issue, it is possible to add options for traces. For instance, if from a SIP device you are not able to dial an ARS prefix, you can add “ars=on” on the actdbg. The Call Handling traces must be adapted to the issue.

Here an example of trace asked by R&D:

```
(1)OXE> tuner km
(1)OXE> tuner clear-traces
(1)OXE> tuner all=off
(1)OXE> trc init
(1)OXE> actdbg all=off
(1)OXE> tuner +at +tr +xtr +s
(1)OXE> tuner +cpu +cpl
(1)OXE> tuner hybrid=on
(1)OXE> actdbg sip=on csip=on fct=on isdn=on abcf=on ext=on rtp=on cnx=on comp=on voip=on ccdn=on cstarout=on
(1)OXE> mtracer -a -u -g
```

Three actdbg options are linked to SIP:

- sip, corresponding to sip (globals SIP traces).
- csip, corresponding to the SEPLOS terminals.
- nsip, corresponding to NOE-SIP terminals (From R10.0).

11.4.3 Tcpdump / Network traces

The tcpdump or network traces can be done as well, to check if the problem is from the network or the network layer of the CPU. Tcpdump must be run under root account.

The network traces are very usefull when you have issue about one way call, DTMF, FAX, etc…

The tcpdump or network traces must be done simultaneously with the SIPMOTOR and the Call Handling traces.

```
(1)OXE> su root
Password:

[root@OXE tmpd]# tcpdump -s 2000 -w trace.cap
tcpdump: listening on eth0, link-type EN10MB (Ethernet), capture size 2000 bytes
```

Runing the tcpdump with the option “-s 2000” and the option “-w trace.cap” is used to be able to open this trace with wireshark (http://www.wireshark.org/).

Rotating trace can be used with the next syntax:

```
[root@OXE tmpd]# tcpdump -C 10 -w /tmpd/mytcpdump.cap -W 10 -s 2000 &
```

-C corresponds to the size of the file (10 corresponds to 10 Megabytes)
-W corrsponds to the number of files
More options are available.

11.5 Mantenance commands

This chapter explains all the SIP maintenance commands available on OXE.

11.5.1 sip

```
=====================================================================
|   T O O L S   A V A I L A B L E   F O R   S I P   P U R P O S E   |
=====================================================================

        trkstat     : Shows the trunks states in a trunk group
        trkvisu     : Shows the trunks parameters in a trunk group
        sipacces    : Shows the SIP trunk group numbers and the related accesses

        sipgateway  : Shows the main SIP gateway parameters
        sipdump     : Shows the main SIP gateway internal resources

        sipextgw    : Shows the external SIP gateways parameters
        sippool     : Shows the external SIP gateways membership of pools

        sipdict     : Shows the SIP dictionnary records
        sipauth     : Shows the SIP authentification records
        sipregister : Shows the SIP end points IP address registered

csipsets     : Shows the list of configured SIP extension
csipview com : Shows the list of SIP extension in communication
csiprestart  : Reset the dynamic datas (CH + CC) of blocked SIP extension

        sipextusers    : Shows the SIP devices with gateway
```

The command “sip” gives all the commands related to SIP.

11.5.2 trkstat

```
+=============================================================================+
|   S I P   T R U N K   S T A T E     Trunk group number : 10                 |
|                                     Trunk group name   : SIP_local          |
|                                     Number of Trunks   : 62                 |
+-----------------------------------------------------------------------------+
|   Index :     1    2    3    4    5    6    7    8    9   10   11   12   13  |
|   State :     F    F    F    F    F    F    F    F    F    F    F    F    F  |
+-----------------------------------------------------------------------------+
|   Index :    14   15   16   17   18   19   20   21   22   23   24   25   26  |
|   State :     F    F    F    F    F    F    F    F    F    F    F    F    F  |
+-----------------------------------------------------------------------------+
|   Index :    27   28   29   30   31   32   33   34   35   36   37   38   39  |
|   State :     F    F    F    F    F    F    F    F    F    F    F    F    F  |
+-----------------------------------------------------------------------------+
|   Index :    40   41   42   43   44   45   46   47   48   49   50   51   52  |
|   State :     F    F    F    F    F    F    F    F    F    F    F    F    F  |
+-----------------------------------------------------------------------------+
|   Index :    53   54   55   56   57   58   59   60   61   62                 |
|   State :     F    F    F    F    F    F    F    F    F    F                 |
+-----------------------------------------------------------------------------+
|  F:  Free   |    B: Busy   |   Ct: busy Comp trunk   |  Cl: busy Comp link   |
|  WB: Busy Without B Channel|   Cr: busy Comp trunk for RLIO inter-ACT link   |
|  WBD: Data Transparency without chan.| WBM: Modem transparency without chan. |
|  D: Data Transparency      |   M: Modem transparency                        |
+-----------------------------------------------------------------------------+
```

The command “trkstat” + SIP Trunk Group number gives the B channels used on the SIP Trunk group assocaited to a gateway.

11.5.3 trkvisu

```
****************** data in Trunk_Group structure ****************

********        data REMOTE TRUNK
TrunkName         = SIP_local
discrLogId     = -1     ton_a_used = 0          em_repfix = 0
privLine       = 0      reservop    = 0         reserauto = 0
Trunksearchs  = 0       ftranscom   = 0         frondier  = 0
typTrunk :    (6) => T2-SIP
Next_Trunk      = -1    nbdigitsem    = 0               tab_proto = -1
var IPNS = 1
node_number    = 1
network_number = 10
trunk_reg_sig = 0
special_it_par_quantum = 1
cat_restrictionService_in  = 10, cat_restrictionService_out = 10
Priority ===> Level= 0, Mode= 0, Preemption= 0
mpt1343 = 0, callbackTrunkbusy = 1 rerouting = 0
********        data Link
cat_signa = 31  ch_channelb = 1 overflow_it = 1 access_turn = 1 network_mode = 0
+-------------------------------------------------------------------------------------------
|                              ocupjonc for SIP TG
+-------------------------------------------------------------------------------------------
|  SIP Trunk group information on TX side
|  i = 0, min = 0, max = 62
|  (num_crist - num_cpl - num_term) = (19-0-1)
|  last it used  = 0,  monlap = 30,  network_mode = 0  nbr_trunk_created = 62
|  nbr_trunk_busy :  start = 0  arrived = 0  mixed = 0
|  it_reserved    :  start = 0  arrived = 0  mixed = 62
|  it_max_Q0       :  Start = 0  arrived = 0  mixed = 62
|  it_max_Q1       :  Start = 0  arrived = 0  mixed = 0
|  access_level2 = CONNECT2
+-------------------------------------------------------------------------------------------
|  outservice  | res | Busy | nulog |trans|neqtdyn|E64 RN64 EN64| OVPN |  neqph  | adr
+-------------------------------------------------------------------------------------------
| FREE         | no  | free |  5001 |   1 |    -1 |  0   0   0  |  0   |    2314 |SIP Trunk 1
...
| FREE         | no  | free |  5062 |   1 |    -1 |  0   0   0  |  0   |    2376 |SIP Trunk 62
+-------------------------------------------------------------------------------------------
+-------------------------------------------------------------------------------------------
|  SIP Trunk group information on GX side
|  (num_crist - num_cpl - num_term) = (19-0-0)
|  monlap = 29,  mode_reseau = 1  nbr_jonc_cree = 62
|  Trunks from nulog 5063 (neqph  2250) to nulog 5124 (neqph  2312)
+-------------------------------------------------------------------------------------------

index_max = 125 ; nbjonc  = 62
cristal     =    0    1    2    3    4    5    6    7    8    9   10   11
LastTrunkused =    0    0    0    0    0    0    0    0    0    0    0    0
cristal     =   12   13   14   15   16   17   18   19
LastTrunkused =    0    0    0    0    0    0    0   62
LastTrunkUsed common = 0        idx_rfo = 0     channel_reserv = 0
dert0_used = 0  dert0mixt_used = 0      dert0wo_used = 0

********        data TRUNK_LOCAL
a_paying    = 0   Trunkdisa     = 0     itpermnt   = 1  trans_num = 0
tr_q23      = 0   reach_boss = 0        secretcode = 0  ach_film  = 1
accesscode    = 0         gp_d_Hold   = 0       categ_ptt  = 31 blf etat  = 1
entity_nr   = 0   nb_digit_used  = 0    Trunkdissu     = 0      dto_about = 0
reused_channelb = 0       number_to_be_added :          .
mode_ddi    = 0   refptt = /      /                     nbchminp  = 0
x25used     = 0
vpnRate     = 50         vpnCostLimit  = 0      immTrkForVpn = 1
businessPercent = 0      nbACDCall     = 0      tax_nds      = 1
send_prog = 1    ip_qual_prof = Profile #1
t2spec      = S_SIP      compression_type = 0 (0: Default [ie : G729], 1 : G711)
d_channel_hyb = 0
```

The information given are the same compares to a “normal” T2 access, this command can be usable to find the equipment of a SIP Trunk Group, or the “neqt”.

A SIP Trunk group has two “sides”, the TX (USER) and GX (NETWORK). When a call is done on a SIP Trunk Group, the call is leaving on the SIP TG and comes back on the same SIP TG, it is like an internal SIP loop.

11.5.4 sipaccess

```
+-------------------------------------------------------------------------+
|   1    |                    SIP Trunk Group Access                        |
+-------------------------------------------------------------------------+
| TG Nb  |      10      |      12      |      11      |     186      |     187      |
|        |              |              |              |              |              |
| Access | User - Net   | User - Net   | User - Net   | User - Net   | User - Net   |
+-------------------------------------------------------------------------+
|    1   |   30 - 29    |   33 - 32    |   35 - 34    |   37 - 36    |   39 - 38    |
|    2   |    . . .     |   41 - 40    |    . . .     |    . . .     |    . . .     |
|    3   |    . . .     |    . . .     |    . . .     |    . . .     |    . . .     |
|    4   |    . . .     |    . . .     |    . . .     |    . . .     |    . . .     |
|    5   |    . . .     |    . . .     |    . . .     |    . . .     |    . . .     |
|    6   |    . . .     |    . . .     |    . . .     |    . . .     |    . . .     |
|    7   |    . . .     |    . . .     |    . . .     |    . . .     |    . . .     |
|    8   |    . . .     |    . . .     |    . . .     |    . . .     |    . . .     |
|    9   |    . . .     |    . . .     |    . . .     |    . . .     |    . . .     |
|   10   |    . . .     |    . . .     |    . . .     |    . . .     |    . . .     |
|   11   |    . . .     |    . . .     |    . . .     |    . . .     |    . . .     |
|   12   |    . . .     |    . . .     |    . . .     |    . . .     |    . . .     |
|   13   |    . . .     |    . . .     |    . . .     |    . . .     |    . . .     |
|   14   |    . . .     |    . . .     |    . . .     |    . . .     |    . . .     |
|   15   |    . . .     |    . . .     |    . . .     |    . . .     |    . . .     |
|   16   |    . . .     |    . . .     |    . . .     |    . . .     |    . . .     |
+-------------------------------------------------------------------------+
```

The command “sipacces” gives the access numbers used for each SIP TG.

In that case, for the TG number 10 with 2 accesses managed, the OXE uses the accesses 30 for TX and 29 for GX, these accesses numbers can be found with the command “trkvisu” (search for “monlap”).

In that case, for the TG number 12 with 4 accesses managed, the OXE uses the accesses 33 and 41 for TX then 32 and 40 for GX.

11.5.5 sipgateway

```
+-------------------------------------------------------------------+
|                          SIP Gateway                              |
+-------------------------------------------------------------------+
Machin name            : oxe-ov
IP Address             : 172.27.142.53

Subnetwork number      : 10
SIP Trunk Group        : 10

DNS Informations :
DNS local domain name  : alcatel.fr

+-------------------------------------------------------------------+
|                       Trusted IP Address List                     |
+-------------------------------------------------------------------+

+-------------------------------------------------------------------+
|                     Quaranted IP Address List                     |
+-------------------------------------------------------------------+
```

The command “sipgateway” gives the information about the local SIP configuration.

The next information are diplayed:

- Machine name corresponds to the “nodename” managed under netadmin.
- IP address corresponds to the main IP address of the main CPU.
- Subnetwork number correponds to the network associated to the local SIP gateway.
- SIP Trunk Group correponds to the SIP TG associated to the local SIP gateway.
- DNS local domain name correponds to the DNS suffix managed on the local SIP gateway.
- Trusted IP Addresses List corresponds to the IP addresses managed on the “Trusted IP Addesses”
- Quaranted IP Addresses List corresponds to the IP addresses managed on the “Quarantined IP addresses”.

11.5.6 sipdump

The sipdump tool gives some information about SIP calls and the SIP gateway. It’s useful in order to know which states have the SIP calls, which are handled by the SIP gateway, to release a call, which is inactive, etc…
It allows defining some filters in order to display the traces of SIP calls, according to SIP calls characteristics (“From”, “To”, “P_Asserted”, “Request URI” headers).

Activation:

- Set a trace level very low (set by motortrace – lowest trace level by motortrace 0), and disable filters.
- Run the “traced &” command.
- Run the command “sipdump”.
- For better view, run “sipdump” and “traced” in different telnet sessions.

A “Call” corresponds to a SIP voice call, but also for a subscription, notify, etc…
Sometimes, choices must be done twice to get the outputs.

R10.x

```
        SIP Gateway resources menu

 1 - Dump the gateway management datas
 2 - Dump a call
 3 - Display the number of calls
 4 - Display the calls-neqt mapping
 5 - Display the calls list
 6 - Display the detailed calls list
 7 - Release a call
 8 - Display subscription list
 9 - Display calls through a gateway
10 - Display calls in a trunk group
11 - SIP traces filters
12 - Display registred users
13 - Display CPU-SSM connections
14 - Display memory allocation
15 - Display IP cache from ext gw
 0 - Exit
```

R9.1

```
        SIP Gateway resources menu

 1 - Dump the gateway management datas
 2 - Dump a call
 3 - Display the number of calls
 4 - Display the calls-neqt mapping
 5 - Display the calls list
 6 - Release a call
 7 - Display subscription list
 8 - Display calls through a gateway
 9 - Display calls in a trunk group
10 - SIP traces filters
 0 - Exit
```

- 1 – Dump the gateway management datas :

```
Wed Jan  4 14:48:42 2012      Gateway Management Datas
Wed Jan  4 14:48:42 2012 ----------------------------------------
Wed Jan  4 14:48:42 2012
Wed Jan  4 14:48:42 2012 Use of licences               : Yes
Wed Jan  4 14:48:42 2012 Number of initial licenses    : 20
Wed Jan  4 14:48:42 2012 Number of available licences  : 15
Mon Jun  4 12:48:42 2012 Number of initial Tls licenses   : 5
Mon Jun  4 12:48:42 2012 Number of available Tls licences : 5
Wed Jan  4 14:48:42 2012
Wed Jan  4 14:48:42 2012 Main server                   : Yes
Wed Jan  4 14:48:42 2012 Degraded mode                 : Yes
```

- Use of licenses means that the OXE is using SIP, license point of view.
- Number of initial licenses corresponds to the number of licenses bought.
- Number of available licenses corresponds to the number of licenses remaining. The difference with the Number of initial licenses give the number of licenses used when this choice is done.
- Number of initial Tls licenses corresponds to the number of licenses bought for TLS.
- Number of available Tls licenses corresponds to the number of licenses remaining for TLS. The difference with the Number of initial Tls licenses give the number of licenses for TLS used when this choice is done.
- Main server gives the role of the CPU where you run the “sipdump” command.
- Degraded mode is used when the SIPMOTOR reaches a threshold of SIP contexts treatment, in that case, the SIPMOTOR switches in degraded mode to reject all the incoming SIP messages by a 503 response, with a "Retry-After" header, is sent to the UAC. This is used to avoid SIPMOTOR crash.

- 2 – Dump a call

Enter the “Neqt” of the SIP equipment + “Dialogid”, to know them, use the choice 4 before.

```
1325686751 -> Wed Jan  4 15:18:56 2012 ----------------------------------------
Wed Jan  4 15:18:56 2012      Call Dump
Wed Jan  4 15:18:56 2012 ----------------------------------------
Wed Jan  4 15:18:56 2012
Wed Jan  4 15:18:56 2012 Neqt            : 968-1
Wed Jan  4 15:18:56 2012 Call ID         : 4f7d5f18a41e48012739fa6565a76e41@172.27.143.186
Wed Jan  4 15:18:56 2012 Current state   : COMPLETED_STATE
Wed Jan  4 15:18:56 2012 From            : sip:32000@172.27.143.186;user=phone
Wed Jan  4 15:18:56 2012 To              : sip:32001@172.27.143.186;user=phone
Wed Jan  4 15:18:56 2012 External VM:    : FALSE
Wed Jan  4 15:18:56 2012 Sip Device:     : FALSE
Wed Jan  4 15:18:56 2012 Ext. Gateway    : Not used
Wed Jan  4 15:18:56 2012 Session Timer   : INVITE method
Wed Jan  4 15:18:56 2012 ----------------------------------------
Wed Jan  4 15:19:07 2012 ----------------------------------------
Wed Jan  4 15:19:07 2012      Neqt - Call mapping
Wed Jan  4 15:19:07 2012 ----------------------------------------
Wed Jan  4 15:19:07 2012
Wed Jan  4 15:19:07 2012 Active Calls (1 / 1)
Wed Jan  4 15:19:07 2012   Eqt =   968 dialogId = 1 <-> Call ID = 4f7d5f18a41e48012739fa6565a76e41@172.27.143.186
Wed Jan  4 15:19:07 2012       State = COMPLETED_STATE
Wed Jan  4 15:19:07 2012
Wed Jan  4 15:19:07 2012
Wed Jan  4 15:19:07 2012 Unactive Calls (0 / 1)
Wed Jan  4 15:19:07 2012 ----------------------------------------
```

- The “Current state”corresponds to the status of the call:
  - PROCEEDING_STATE : the call is in progress (ringing for instance).
  - COMPLETED_STATE : the call is established.
  - TERMINATED_STATE : the call is finished.

- From and To correspond to the caller and the callee.

- External VM : False means that it is not an external SIP Voice mail.

- Sip Device: False means a SIP extension user (SEPLOS).

- Ext. Gateway correponds to the external SIP gateway used for the call.

- Session Timer corresponds to the method used for it according to the local SIP gateway management:
  - UPDATE Method: use UPDATE message to refresh the session.
  - INVITE method: use RE_INVITE message to refresh the session.
- Active Calls correponds to the SIP calls established
  - Only COMPLETED_STATE is visible.

- Unactive Calls corresponds to the SIP calls over or in progress:
  - Unactive + PROCEEDING_STATE, corresponds to a SIP call in progress.
  - Unactive + TERMINATED_STATE, corresponds to a SIP call over, but its SIP context is still present on the SIPMOTOR. The maximum duration of the context in the SIPMOTOR is 32 seconds, during this period, the SIPMOTOR will delete it. If the SIP call context is still present after this delay, the SIPMOTOR will not be able to remove it by itself, a restart of the SIPMOTOR must be done.

When a restart of the SIPMOTOR is done, all the SIP call context will be lost, that means that the calls are not anymore known by the SIPMOTOR.

- 3 – Display the number of calls

```
1325752599 -> Thu Jan  5 09:36:39 2012 stack data.

Thu Jan  5 09:36:39 2012 ==========
Thu Jan  5 09:36:39 2012 Calls :               1 current (4 max) / 59052 total
Thu Jan  5 09:36:39 2012 Dialogs :             1 current (6 max) / 59083 total
Thu Jan  5 09:36:39 2012 Transactions :        1 current (6 max) / 59240 total
Thu Jan  5 09:36:39 2012 Requests :            1 current / 59301 total
Thu Jan  5 09:36:39 2012 Response :            0 current / 309 total
Thu Jan  5 09:36:39 2012 DNS requests :        0 current (0 max) / 0 total / 0 foundInCache / 0 failed / 0
totalPutinBlackList…
```

  - Corresponds to the number of SIP calls, but also SIP dialogs, SIP transactions, etc…

- 4 - Display the calls-neqt mapping.

```
Thu Jan  5 10:19:59 2012 ----------------------------------------
Thu Jan  5 10:19:59 2012      Neqt - Call mapping
Thu Jan  5 10:19:59 2012 ----------------------------------------
Thu Jan  5 10:19:59 2012
Thu Jan  5 10:19:59 2012 Active Calls (1 / 1)
Thu Jan  5 10:19:59 2012   Eqt =   968 dialogId = 2 <-> Call ID = 63aa134016f94865c97c22a4e6a0c20c@172.27.143.186
Thu Jan  5 10:19:59 2012       State = COMPLETED_STATE
Thu Jan  5 10:19:59 2012
Thu Jan  5 10:19:59 2012
Thu Jan  5 10:19:59 2012 Unactive Calls (0 / 1)
Thu Jan  5 10:19:59 2012 ----------------------------------------
```

- Corresponds to the Active and Unactive calls present on SIPMOTOR, for the sipdump choice 2, it is necessary to have the “Neqt” and the “dialogid”, here we have them for each call.

- 5 - Display the calls list.

```
Thu Jan  5 10:25:54 2012 ------------------------------------------
Thu Jan  5 10:25:54 2012      List of Calls
Thu Jan  5 10:25:54 2012 ------------------------------------------
Thu Jan  5 10:25:54 2012
Thu Jan  5 10:25:54 2012 Active Calls (1 / 1)
Thu Jan  5 10:25:54 2012   Call ID = 63aa134016f94865c97c22a4e6a0c20c@172.27.143.186
Thu Jan  5 10:25:54 2012       State = COMPLETED_STATE
Thu Jan  5 10:25:54 2012
Thu Jan  5 10:25:54 2012
Thu Jan  5 10:25:54 2012 Unactive Calls (0 / 1)
Thu Jan  5 10:25:54 2012 ------------------------------------------
```

- List the Active and Unactive SIP calls on the SIPMOTOR. Recommended in case of licence consumming issue.

- 6 - Display the detailed calls list.

```
Thu Jan  5 10:29:41 2012      Detailed list of Calls from Stack
Thu Jan  5 10:29:41 2012 ------------------------------------------
Thu Jan  5 10:29:41 2012  102 [CCallManager] Dump - 1 CCall instance(s)
[1137] Call ID : 63aa134016f94865c97c22a4e6a0c20c@172.27.143.186
CCall 1137
Call-ID              : 63aa134016f94865c97c22a4e6a0c20c@172.27.143.186
isClosed             : no
onlyInitialDialog    : no

  ============================================================
  InitialDialog client :
  --------------------
  CDialog 1537
  isClosed       : no
  isProxy        : no
  isRouted       : no
  State          : Initial
  Initial method : INVITE
  Session-Timer  :
                   isProxy         : no
                   supported       : I support
                   Min-SE          : 900
                   Session-Expires : 1800
                   Refresher       : I refresh
                   Warning timer   : stopped
                   Session timer   : stopped
                   Refresh method  :
  Route set      : Contact : sip:32001@172.27.141.210:36128;rinstance=98cedca3f085d785
  Messages :
  ----------------------------------------
            out:INVITE [2012/01/05 10:19:54 CET]
            in:180 (INVITE) [2012/01/05 10:19:54 CET]
            in:200 (INVITE) [2012/01/05 10:19:55 CET]
  ----------------------------------------
  ------------------------------------------------------------
    Transactions :
    ------------
    CTransaction 2138
    State               : Proceeding
    isClient            : yes
    isCancelable        : no
    isRouted            : no
    isProxy             : no
    Initial request     : INVITE (38)
    Last response       : 180 (6)
    Final response      : None
    Ack request         : None
    Timers in progress : None
    ------------------------------------------------------------
```

- This choice is used to view the different exchanges details for the SIP transactions.
- For each transaction, we have 3/4 groups of information (3for call in progress, 4 for established/closed):

    - SIP call information with the Call ID and the state of the call:
        - Closed (isClosed= yes)
        - In progress (onlyInitialDialog=yes)
        - Established (isClosed= no and onlyInitialDialog=no)
    - InitialDialog client:
        - This part corresponds to the information on the SIP message received or sent to establish a SIP transaction (INVITE, SUBSCRIBES, etc…).
    - Transaction:
        - This part corresponds to the status of the transaction itself (type of transaction, last message, etc…).
    - Dialogs:
        - This part corresponds to the dialog information.

- 7 - Release a call.
    - Enter the “Neqt” number and the “DialogId”, use the choice 4 to find them.

```
Thu Jan  5 12:05:45 2012 ----------------------------------------
Thu Jan  5 12:05:45 2012      Neqt - Call mapping
Thu Jan  5 12:05:45 2012 ----------------------------------------
Thu Jan  5 12:05:45 2012
Thu Jan  5 12:05:45 2012 Active Calls (1 / 1)
Thu Jan  5 12:05:45 2012   Eqt =   968 dialogId = 1 <-> Call ID = 94e2d5c0e0bd3549d61fbaf643e4779a@172.27.143.186
Thu Jan  5 12:05:45 2012       State = COMPLETED_STATE
Thu Jan  5 12:05:45 2012
Thu Jan  5 12:05:45 2012
Thu Jan  5 12:05:45 2012 Unactive Calls (0 / 1)
Thu Jan  5 12:05:45 2012 ----------------------------------------
Thu Jan  5 12:05:51 2012 ALARM: [receiveSuccessfulEvent] Call: 94e2d5c0e0bd3549d61fbaf643e4779a@172.27.143.186
eqt: 968 TERMINATED_STATE failed to emit a Successful message.
Thu Jan  5 12:05:51 2012 ALARM: CPU main
```

- An incident 5816 is seen on the OXE and the alarm is visible on the sipalarm files.

- 8 - Display subscription list.

```
Thu Jan  5 12:11:33 2012 ----------------------------------------
Thu Jan  5 12:11:33 2012 sipmotor Subscription Map
Thu Jan  5 12:11:33 2012        key      32001@172.27.143.186@message-summary
Thu Jan  5 12:11:33 2012        call no  1153
Thu Jan  5 12:11:33 2012        call Id  NTUyZjA1ZmFiYTQ1MDI3N2U2ZTE1NzFkY2ZjZmM2MmQ.
Thu Jan  5 12:11:33 2012        delay    3600
Thu Jan  5 12:11:33 2012 ----------------------------------------
```

- The subscription can be used in case of voice mail, for instance to be able to be notified if a message has been deposited on the voice mailbox.

- 9 - Display calls through a gateway.
    - Enter the External Gateway number.

```
Thu Jan  5 13:41:14 2012 ----------------------------------------
Thu Jan  5 13:41:14 2012 Call ID         : 763eb45a1543ce4b31174e4081285074@172.27.143.186
Thu Jan  5 13:41:14 2012 Current state   : COMPLETED_STATE
Thu Jan  5 13:41:14 2012 From            : sip:32000@toto;user=phone
Thu Jan  5 13:41:14 2012 To              : sip:31002@oxe-ov.alcatel.fr;user=phone
Thu Jan  5 13:41:14 2012 Session Timer   : UPDATE method
Thu Jan  5 13:41:14 2012 ----------------------------------------
Thu Jan  5 13:41:14 2012 Number of Calls through this Gateway (151) : 1  (Active calls: 1)
```

- 10 - Display calls in a trunk group.
  - Enter the SIP trunk group number (ISDN or ABCF).

```
Trunk Group Number : 10


                Display of trunk groups Menu

         1 - Display calls through any one gateway using the trunk group(10)
         2 - Display calls through all the gateways using the trunk group(10)
         0 - Previous menu
```

  - Select 1 or 2, if 1 enter the SIP gateway number (0 to 999).

```
Thu Jan  5 13:49:50 2012 ------------------------------------------
Thu Jan  5 13:49:50 2012 Call ID         : 4661cf1f0940acda70ed8302c8050f79@172.27.143.186
Thu Jan  5 13:49:50 2012 Current state   : COMPLETED_STATE
Thu Jan  5 13:49:50 2012 From            : sip:32000@toto;user=phone
Thu Jan  5 13:49:50 2012 To              : sip:31002@oxe-ov.alcatel.fr;user=phone
Thu Jan  5 13:49:50 2012 Gateway         : 151
Thu Jan  5 13:49:50 2012 Session Timer   : UPDATE method
Thu Jan  5 13:49:50 2012 ------------------------------------------
Thu Jan  5 13:49:50 2012 Number of Calls in this Trunkgroup (10) : 1 (Active calls: 1)
Thu Jan  5 13:49:50 2012 ------------------------------------------
```

- 11 - SIP traces filters.

This functionality allows setting up to five filters on SIP gateway calls. A filter is composed of the following elements:

- Filter string: String to search into the SIP calls headers the user wants to trace.
- From Field: If the field is set true, the user traces the SIP calls according to the content of “From” header. In this case, if the SIP call ‘From’ header contains the filter string defined for the filter, the SIP call will be traced.
- To Field: If the field is set true, the user traces the SIP calls according to the content of “To” header.
- P_Asserted field: If the field is set true, the user traces the SIP calls according to the content of “P_Asserted” header.
- Request-URI field: If the field is set true, the user traces the SIP calls according to the content of the Request URI.

Display conditions:

- SIP call traces will be displayed if the SIP call matches at least one of the five filters of the array.
- A SIP call matches to a filter if it fills one of the conditions of the filter.

```
            SIP traces filters menu

     1 - Display the traces filters
     2 - Add a traces filter
     3 - Update a traces filter
     4 - Remove a traces filter
     5 - Remove all traces filters
     0 - Previous menu
```

- 1 - Display the traces filters.

```
-------------------------------------------------------------------------------
| Nb | Filter                          | From |  To  | P_Asserted | Request URI |
|-----------------------------------|------|------|------------|-------------|
| 1  | ...                             | ...  | ...  |    ...     |     ...     |
|-----------------------------------|------|------|------------|-------------|
| 2  | ...                             | ...  | ...  |    ...     |     ...     |
|-----------------------------------|------|------|------------|-------------|
| 3  | ...                             | ...  | ...  |    ...     |     ...     |
|-----------------------------------|------|------|------------|-------------|
| 4  | ...                             | ...  | ...  |    ...     |     ...     |
|-----------------------------------|------|------|------------|-------------|
| 5  | ...                             | ...  | ...  |    ...     |     ...     |
-------------------------------------------------------------------------------
```

- 2 - Add a traces filter.

```
String to filter ? (31 car. max) :

From field ? (y/n) :

To field ? (y/n) :

P_Asserted Field ? (y/n) :

Request URI field ? (y/n) :
```

  - Enter which information to filter (the filters are not case sensitive), and manage on each field to use it or not.

```
-------------------------------------------------------------------------------
| Nb | Filter                          | From |  To  | P_Asserted | Request URI |
|-----------------------------------|------|------|------------|-------------|
| 1  | alcatel-lucent.com              | Yes  | Yes  |    Yes     |     Yes     |
|-----------------------------------|------|------|------------|-------------|
| 2  | genesys.com                     | Yes  | Yes  |    Yes     |     Yes     |
|-----------------------------------|------|------|------------|-------------|
| 3  | ...                             | ...  | ...  |    ...     |     ...     |
|-----------------------------------|------|------|------------|-------------|
| 4  | ...                             | ...  | ...  |    ...     |     ...     |
|-----------------------------------|------|------|------------|-------------|
| 5  | ...                             | ...  | ...  |    ...     |     ...     |
-------------------------------------------------------------------------------
```

- 3 - Update a traces filter.
  - Enter the filter number, in this case, only the filter 1 is managed.

```
From field ? (y/n) : y

To field ? (y/n) : y

P_Asserted Field ? (y/n) : n

Request URI field ? (y/n) : y
```

- The filter string can not be modified, only on witch field it is used.

```
-------------------------------------------------------------------------------
| Nb | Filter                          | From |  To  | P_Asserted | Request URI |
|-----------------------------------|------|------|------------|-------------|
| 1  | alcatel-lucent.com              | Yes  | Yes  |    No      |     Yes     |
|-----------------------------------|------|------|------------|-------------|
| 2  | genesys.com                     | Yes  | Yes  |    Yes     |     Yes     |
|-----------------------------------|------|------|------------|-------------|
| 3  | ...                             | ...  | ...  |    ...     |     ...     |
|-----------------------------------|------|------|------------|-------------|
| 4  | ...                             | ...  | ...  |    ...     |     ...     |
|-----------------------------------|------|------|------------|-------------|
| 5  | ...                             | ...  | ...  |    ...     |     ...     |
-------------------------------------------------------------------------------
```

- 4 - Remove a traces filter.

```
-------------------------------------------------------------------------------
| Nb | Filter                          | From |  To  | P_Asserted | Request URI |
|-----------------------------------|------|------|------------|-------------|
| 1  | alcatel-lucent.com              | Yes  | Yes  |    No      |     Yes     |
|-----------------------------------|------|------|------------|-------------|
| 2  | genesys.com                     | Yes  | Yes  |    Yes     |     Yes     |
|-----------------------------------|------|------|------------|-------------|
| 3  | ...                             | ...  | ...  |    ...     |     ...     |
|-----------------------------------|------|------|------------|-------------|
| 4  | ...                             | ...  | ...  |    ...     |     ...     |
|-----------------------------------|------|------|------------|-------------|
| 5  | ...                             | ...  | ...  |    ...     |     ...     |
-------------------------------------------------------------------------------

Filter to remove (0:Previous menu) :
```

- Enter the filter number, only this one will be removed (1 for instance).

```
-------------------------------------------------------------------------------
| Nb | Filter                          | From |  To  | P_Asserted | Request URI |
|-----------------------------------|------|------|------------|-------------|
| 1  | ...                             | ...  | ...  |    ...     |     ...     |
|-----------------------------------|------|------|------------|-------------|
| 2  | genesys.com                     | Yes  | Yes  |    Yes     |     Yes     |
|-----------------------------------|------|------|------------|-------------|
| 3  | ...                             | ...  | ...  |    ...     |     ...     |
|-----------------------------------|------|------|------------|-------------|
| 4  | ...                             | ...  | ...  |    ...     |     ...     |
|-----------------------------------|------|------|------------|-------------|
| 5  | ...                             | ...  | ...  |    ...     |     ...     |
-------------------------------------------------------------------------------
```

- 5 - Remove all traces filters.
  - If you choose this option, all the filters will be removed.

Example: The traces must be done when **alcatel-lucent.com** is present on the “To” or the “From” field and/or **genesys.com** on the “From” or the “P_Asserted” fields .

The result is next:

Nb	Filter	From	To	P_Asserted	Request URI
1	alcatel-lucent.com	Yes	Yes	...	...
2	genesys.com	Yes	...	Yes	...
3	...	...	...	...	...
4	...	...	...	...	...
5	...	...	...	...	...

When you will make a SIP trace (motortrace + traced), the OXE will display the SIP exchanges and information according to the filter management.

If you run the motortrace command and if a filter is managed, the next message will be displayed:

```
motortrace (v5.2.0) verbosity = 00800004
The sipmotor traces level can not be changed
        because some traces filters are set.
Please, remove them (with sipdump) before updating the traces level.
```

Do not forget to remove all the filters after use.

- 12 - Display registred users.

```
Thu Jan  5 15:12:34 2012 ------------------------------------------
Thu Jan  5 15:12:34 2012      Detailed list of Registred users
Thu Jan  5 15:12:34 2012 ------------------------------------------
Thu Jan  5 15:12:34 2012
Thu Jan  5 15:12:34 2012 ************************************************
[CServRegistrar] Dump local registrar base
Address of record : 32003
        contact : sip:32003@172.27.141.210:46470, , 1611sec, 0.5
------------------------------------------------
Registrar statistics :
        Number of users recorded : 1
        Number of users having multiple contacts : 0
        Number of contacts using UDP transport : 1
        Number of contacts using TCP transport : 0
************************************************

Thu Jan  5 15:12:34 2012 ------------------------------------------
```

- Comparing to the “sipregister” command, here there is statistics about the Registrar.

- Number of users recorded corresponds to the number of SIP equipments registered on the OXE Registrar.
- Number of users having multiple contacts corresponds to the SIP equipments with multiple contacts, used in case of forking.
- Number of contacts using UDP transport corresponds to the number of contact using UDP.
- Number of contacts using TCP transport corresponds to the number of contact using TCP.

11.5.7 sipextgw

This command is used with options:

- sipextgw -l gives the external SIP gateways created and their states.

```
=====================================================================
| R E G I S T E R E D   S I P   E X T E R N A L   G A T E W A Y S  |
=====================================================================

        IN SERVICE SIP external gateways list :
        186

        OUT OF SERVICE SIP external gateways list :
        187
```

Here the external SIP gateway 186 is “in service” and the external SIP gateway 187 is “out of service”.

- sipextgw -g “external gateway number” gives the configuration of this external SIP gateway.

```
=====================================================================
|       S I P   E X T E R N A L   G A T E W A Y   Nb 186           |
=====================================================================
        Gateway Name              : SIMUL_SIP_ABCF
        Gateway Type              : Standard type
        State                     : IN SERVICE
        Belong to pool number     : -1
        Use trunk group number    : 186 (ABC-F)
        Remote domain             : 172.27.143.186
        Port number               : 5060
        Transport                 : UDP
        SRTP                      : RTP only
        Prack                     : NO
        Clir                      : YES
        SIP info enable           : NO
        Authentication method     : NONE
        SDP in 180 messages       : NO
        Payload                   : 97
        Outgoing username         :
        Outgoing password         : *****
        Incoming username         :
        Incoming password         : *****
        Local domain name         :
        Local user name           :
        Realm name                :
        Outbound proxy            :
        Supervision timer         : 0
        Registration timer        : 0
        DNS type                  : DNS A
        Primary DNS IP address    : 000.000.000.000
        Secondary DNS IP address  : 000.000.000.000
        PCS IP address            : 000.000.000.000
        Retransmission number
        of REGISTER/OPTIONS       : 2
        Service route index       : -1
        P-Asserted-ID             : FALSE
        TrustedPAssIDHeader       : TRUE
        TrustedFromHeader         : FALSE
        Diversion Info to
        provide through           : History Info
```

This command is used to get a quick view of the configuration given to this exteranl SIP gateway.

- sipextgw -s “external gateway number” gives information if the external SIP gateway is used on a “Command table” (ARS) or/and a “Routing Number Table”.

```
=====================================================================
|       E X T E R N A L   G A T E W A Y   Nb 187   A R E A S       |
=====================================================================

        Found in ARS ==> dialling command table number : 187

        Not found in ROUTING tables
```

Here the external SIP gateway 187 is used on the “command table” 187.

```
=====================================================================
|       E X T E R N A L   G A T E W A Y   Nb 186   A R E A S       |
=====================================================================

        Not found in ARS tables

        Found in ROUTING table number : 12
```

Here the external SIP gateway 186 is used on the “Routing table” 12.

11.5.8 sippool

this command is used to the external SIP gateways associated to the same pool.

```
+-----------------------------------+
|          |             |          |
| pool Nb  |   GW 1      |   GW 2   |
|          |             |          |
+-----------------------------------+
|    00    |    187 OOS  | L  186   |
|    01    |   . . .     |  . . .   |
|    02    |   . . .     |  . . .   |
|    03    |   . . .     |  . . .   |
...
|    296   |   . . .     |  . . .   |
|    297   |   . . .     |  . . .   |
|    298   |   . . .     |  . . .   |
|    299   |   . . .     |  . . .   |
+-----------------------------------+
```

Here the external SIP gateways 186 and 187 are in the same pool, the pool number 0.

- "L" shows the latter gateway used from the pool.
- "OOS" means that the related gateway is OUT OF SERVICE.

11.5.9 sipdict

This command is used with options:

- sipdict -l is used to list the sip users.

```
SIP DICTIONNARY, dim = 128, nb records = 16

+----------+----+-----------------------------------------+----+-----+------+------+------+-----+
|          |    |                                         |    |     |      |      | Ext. |     |
| mcdu     | i  | url                                     |Type| Org | idx1 | idx2 |  gw  | Reg |
+----------+----+-----------------------------------------+----+-----+------+------+------+-----+
|    31020 |  0 |                  31020@          oxe-ov |  3 |   1 |   12 |    0 |  --  | --  |
|    31021 |  0 |                  31021@          oxe-ov |  3 |   1 |   15 |    1 |  --  | --  |
|    39002 |  0 |                  39002@         oxeb-ov |  3 |   2 |    3 |    4 |  --  | --  |
|    31853 |  1 |                  31853@   opentouch-ov |  2 |   1 |   14 |   10 |   1  | No  |
|    31022 |  0 |                  31022@          oxe-ov |  3 |   1 |    1 |   11 |  --  | --  |
|    31026 |  0 |                  31026@          oxe-ov |  3 |   1 |    4 |    9 |  --  | --  |
|    31040 |  0 |                  31040@          oxe-ov |  2 |   1 |   10 |    5 |  --  | --  |
|    31041 |  0 |                  31041@          oxe-ov |  2 |   1 |   11 |   13 |  --  | --  |
|    31028 |  0 |                  31028@          oxe-ov |  3 |   1 |    9 |    8 |  --  | --  |
|    31025 |  0 |                  31025@          oxe-ov |  2 |   1 |    5 |    6 |  --  | --  |
|    31023 |  0 |                  31023@          oxe-ov |  3 |   1 |   13 |    7 |  --  | --  |
|    31024 |  0 |                  31024@          oxe-ov |  2 |   1 |    8 |   12 |  --  | --  |
|    31852 |  0 |                  31852@          oxe-ov |  1 |   1 |    0 |    3 |  --  | --  |
|    31027 |  0 |                  31027@          oxe-ov |  3 |   1 |    7 |   15 |  --  | --  |
|    31854 |  0 |                  31854@          oxe-ov |  3 |   1 |    6 |   14 |  --  | --  |
|    31853 |  0 |                  31853@          oxe-ov |  2 |   1 |    2 |    2 |  --  | --  |
+----------+----+-----------------------------------------+----+-----+------+------+------+-----+
```

For each user directory number,the next information are present:

- the “mcdu” corresponds to the directory number of the SIP user.
- the “i” is used to see if the SIP user is linked to an external SIP gateway (0=no, 1=yes)
- the “dim” corresponds to the size of the dictionnary, if the number of the SIP users created is greater than 128, the OXE add one more 128 to have 256, if the number is greater than 256, the OXE add one more 128 to have 384, etc...the maximum is 128*80.
- the “url” corresponds to the SIP url known by the OXE.
  - the user 39002 is from another node (oxeb-ov).
- the “type” corresponds a SIP device or SIP extension:
  - 1 is an external SIP voice mail.
  - 2 is SIP device.
  - 3 is SIP extension.
- the “org” corresponds to the origin node.
- the “idx1” and idx2” are assigned to the SIP users during creation and used internally.
- the “Ext.gw” is used in case of “Open Touch” configuration, only for SIP device.
  - The user 31025 is using it, and it is know by the OXE by 31853@oxe-ov and 31853@172.27.143.186 on OXE.
  - 172.27.143.186 is the “SIP Remote domain” managed on the external SIP gateway 186.
- the “Reg” is used to see if the user is registered on the external SIP gateway.

- sipdict -i is used to list the sip users by using the “idx1” (or “pos”).

```
SIP DICTIONNARY, dim = 128, nb records = 16

+------+----------+----+-------------------------------------------------------+----+------+-----+
|      |          |    |                                                       |    | Ext. |     |
| pos  | mcdu     | i  | url par index                                         |Type|  gw  | Reg |
+------+----------+----+-------------------------------------------------------+----+------+-----+
|   12 |    31852 |  0 |                       31852@                  oxe-ov |  1 |  --  | --  |
|   15 |    31853 |  0 |                       31853@                  oxe-ov |  2 |  --  | --  |
|    3 |    31853 |  1 |                       31853@          172.27.143.186 |  2 |  186 | No  |
|   14 |    31854 |  0 |                       31854@                  oxe-ov |  3 |  --  | --  |
...
```

- sipdict -v is used to list the sip users by using the “idx2”.

- sipdict -n “directory number of the SIP user” is used to display the url associated.

```
(101)cpub_ov> sipdict -n 31027

Thu May 31 09:26:14 CEST 2012

URL = 31027@oxe-ov
```

- sipdict -u “url of the SIP user” is used to display the mcdu associated.

```
(101)cpub_ov> sipdict -u 31027 oxe-ov

Thu May 31 09:28:39 CEST 2012

31027@oxe-ov :
             31027
```

Enter the url without the @ but just a space.

11.5.10 sipauth

This command is used with options:

- sipauth -l is used to list the sip users.

```
SIP AUTHENTIFICATION, dim = 128, nb records = 13

+----------+-------------------------------------------------------------+------+
| mcdu     | authentification                                            | idx1 |
+----------+-------------------------------------------------------------+------+
|    31020 |                                        31020 @       0000 |    2 |
|    31021 |                                        31021 @       0000 |   12 |
|    31853 |                                        31853 @       0000 |    1 |
|    31022 |                                        31022 @       0000 |    3 |
|    31026 |                                        31026 @       0000 |    9 |
|    31040 |                                        31040 @       0000 |   10 |
|    31041 |                                        31041 @       0000 |    8 |
|    31028 |                                        31028 @       0000 |    4 |
|    31025 |                                        31025 @       0000 |   11 |
|    31023 |                                        31023 @       0000 |    7 |
|    31024 |                                        31024 @       0000 |    0 |
|    31027 |                                        31027 @       0000 |    6 |
|    31854 |                                        31854 @       0000 |    5 |
+----------+-------------------------------------------------------------+------+
```

For each user directory number,the next information are present:

- the “mcdu” correponds to the directory number of the SIP user.
- the “authentication” corresponds to the user login and user pass for the authentication, to managed on the SIP equipment if needed.
- the “idx1” is assigned to the SIP users during creation and used internaly, same than the one given by the “sipdict” command.

- sipauth -i is used to list the sip users by using the “idx1”.

```
SIP AUTHENTIFICATION, dim = 128, nb records = 13

+------+----------+-----------------------------------------------------------+
| pos  | mcdu     | authentification                                          |
+------+----------+-----------------------------------------------------------+
|    2 |    31020 |                                          31020 @       0000 |
|   12 |    31021 |                                          31021 @       0000 |
|    1 |    31853 |                                          31853 @       0000 |
|    3 |    31022 |                                          31022 @       0000 |
|    9 |    31026 |                                          31026 @       0000 |
|   10 |    31040 |                                          31040 @       0000 |
|    8 |    31041 |                                          31041 @       0000 |
|    4 |    31028 |                                          31028 @       0000 |
|   11 |    31025 |                                          31025 @       0000 |
|    7 |    31023 |                                          31023 @       0000 |
|    0 |    31024 |                                          31024 @       0000 |
|    6 |    31027 |                                          31027 @       0000 |
|    5 |    31854 |                                          31854 @       0000 |
+------+----------+-----------------------------------------------------------+
```

- sipauth -n “directory number of the SIP user” is used to display the user login and user pass.

```
(101)cpub_ov> sipauth -n 31027

Thu May 31 09:36:56 CEST 2012

LOGIN = 31027@0000
```

11.5.11 sipregister

This command is used with options:

- sipregister, without option, display all the SIP and SIPS users registered on registrar.

```
sipregister h       To get help menu.
*********************************************
Dump local registrar base
-------------------------------------------------
Address of record : 31026
contact : sip:31026@172.27.141.210:27836, udp, 502 s
-------------------------------------------------
Address of record : 31022
contact : sip:31022@172.27.141.206, udp, 2867 s
-------------------------------------------------
Address of record : 31853
contact : sip:31853@172.27.143.186, UDP, 319998256 s
-------------------------------------------------
Address of record : 31023
contact : sip:31023@135.118.226.39:1714, udp, 3300 s
-------------------------------------------------
Address of record : 31027
contact : sip:31027@172.27.143.184, udp, 840 s
*********************************************
******   registred user number : 5
*********************************************
```

For each address of record,the next information are present and given by the remote SIP equipment during registration:

- the “contact” corresponds to the SIP address of the SIP equipment with the IP address to locate it.
- the “upd” corresponds to the transport type, tcp can be shown if it is used.
- The “xx s” corresponds to the registration time left.
- If no port number, the OXE will use the port 5060

- sipregister l provides all the SIP users registered on the registrar (option c is used for SIPS users)

```
sipregister h       To get help menu.
*********************************************
Dump local registrar base
---------------------------------------------
Address of record : 31026
contact : sip:31026@172.27.141.210:27836, udp, 502 s
---------------------------------------------
Address of record : 31022
contact : sip:31022@172.27.141.206, udp, 2867 s
---------------------------------------------
Address of record : 31853
contact : sip:31853@172.27.143.186, UDP, 319998256 s
---------------------------------------------
Address of record : 31023
contact : sip:31023@135.118.226.39:1714, udp, 3300 s
---------------------------------------------
Address of record : 31027
contact : sip:31027@172.27.143.184, udp, 840 s
*********************************************
******   registred user number : 5
*********************************************
```

11.5.12 csipsets

This command is used with options:

- csipsets with no option provides all the SIP extension created on OXE.

```
+-----+--------+----------------+---------------+-----+
|Neqt |Number  |Name            |IP address     |State|
+-----+--------+----------------+---------------+-----+
|02054|31020   |MyIc_touch 172.2|         Unused|  HS |
|02055|31027   |OT4135          | 172.27.143.184|  ES |
|02058|31021   |RO31021         |         Unused|  HS |
|02059|31022   |31022           | 172.27.141.206|  HS |
|02061|31026   |31026           | 172.27.141.210|  ES |
|02064|31028   |MyIC_phone      |         Unused|  HS |
|02066|31023   |31023           |         Unused|  HS |
|02068|31854   |31854           |         Unused|  ES |
+-----+--------+----------------+---------------+-----+
|Number of SIP extensions: 00008                      |
+-----------------------------------------------------+
```

For each user directory number,the next information are present:

- the “Neqt” correponds to the equipment number of the SIP extension given during its creation.
- the “Number” corresponds to the directory number of the SIP extension.
- the “Name” corresponds the name of the SIP extension.
- the “IP address” corresponds to the IP address of the SIP equipment associated to this SIP extension, if “Unused” is shown, that means that no SIP equipment is registered for this user.
- the “State” corresponds to the status of the SIP extension:
  - HS means that the user is Out Of Service.
  - ES means that the user is In Service.

The combination of the “IP address” and the “State” gives you more information:

- If the “IP address” is “Unused” and the “State” is ES:
  - the user is created, but no SIP equipment has been registered for this user.
- If the “IP address” is “Unused” and the “State” is HS:
  - the user has been already registered, but not anymore.
- If the “IP address” is full with an IP address and the “State” is HS:
  - the user is registered, but the user is Out Of Service, this can be possible due to the “keep alive” mechanism for SIP extension. After registartion, the SIP extension doesn’t send or answer to the OPTION messages.
- If the “IP address” is full with an IP address and the “State” is ES:
  - the user is registrered and In Service.

o csipsets d “directory number” gives the information only for this user.

```
(101)cpub_ov> csipsets d 31026

Mon Jun  4 14:08:56 CEST 2012
+-----+--------+----------------+---------------+-----+
|Neqt |Number  |Name            |IP address     |State|
+-----+--------+----------------+---------------+-----+
|02061|31026   |31026           | 172.27.141.210|  ES |
+-----+--------+----------------+---------------+-----+
```

o csipsets n “neqt number” gives the information only for this user.

```
(101)cpub_ov> csipsets n 2061

Mon Jun  4 14:09:54 CEST 2012
+-----+--------+----------------+---------------+-----+
|Neqt |Number  |Name            |IP address     |State|
+-----+--------+----------------+---------------+-----+
|02061|31026   |31026           | 172.27.141.210|  ES |
+-----+--------+----------------+---------------+-----+
```

11.5.13 csipview com

Displays all the SIP extension calls.

o No calls present, the display is next:

```
(101)cpub_ov> csipview com

Mon Jun  4 14:10:28 CEST 2012
+-----+--------+----------------+---------------+--------+
|Neqt |Number  |Name            |IP address     |Activity|
+-----+--------+----------------+---------------+--------+
+-----+--------+----------------+---------------+--------+
|Number of SIP extensions in communication: 00000        |
+--------------------------------------------------------+
```

- Calls are present, the display is next:

```
(101)cpub_ov> csipview com

Mon Jun  4 14:13:41 CEST 2012
+-----+--------+----------------+---------------+---------+
|Neqt |Number  |Name            |IP address     |Activity|
+-----+--------+----------------+---------------+---------+
|02061|31026   |31026           | 172.27.141.210|CH-CC   |
+-----+--------+----------------+---------------+---------+
|Number of SIP extensions in communication: 00001        |
+--------------------------------------------------------+
```

For each user directory number,the next information are present:

- the “Neqt” corresponds to the equipment number of the SIP extension given during its creation.
- the “Number” corresponds to the directory number of the SIP extension.
- the “Name” corresponds the name of the SIP extension.
- the “IP address” corresponds to the IP address of the SIP equipment associated to this SIP extension, if “Unused” is shown, that means that no SIP equipment is registered for this user.
- the “Activity” corresponds to the presence of a “Call Control Half Com”. The “Call Control Half Com”is in charge to interface the SIP world to the OXE world.

11.5.14 csiprestart

This command is used with options:

- csiprestart d “directory number” restarts the SIP extension user:

```
(101)cpub_ov> csiprestart d 31026

Mon Jun  4 14:27:09 CEST 2012
```

- csiprestart n “neqt number” restarts the SIP extension user:

```
(101)cpub_ov> csiprestart n 2061

Mon Jun  4 14:27:09 CEST 2012
```

The option -f exist to force the restart if needed

11.5.15 sipextusers (Only in R10.x for Open Touch).

This command is used with options:

- sipextusers without option gives the list of the SIP users associated to an Open Touch:

```
+---------+---------------------+------+----------+
| Number  |Name                 |Ext GW|Registered|
+---------+---------------------+------+----------+
| 60999   |          OXE_ADV_PROF|000001|       Yes|
| 60001   |       Dujardin Loulou|000001|        No|
| 60002   |         Lamy Chouchou|000001|        No|
| 60050   |               Sy Omar|000001|        No|
+---------+---------------------+------+----------+
|Number of SIP USERS: 00004      |
+--------------------------------+
```

- sipextusers -d “directory number” of the SIP device user:

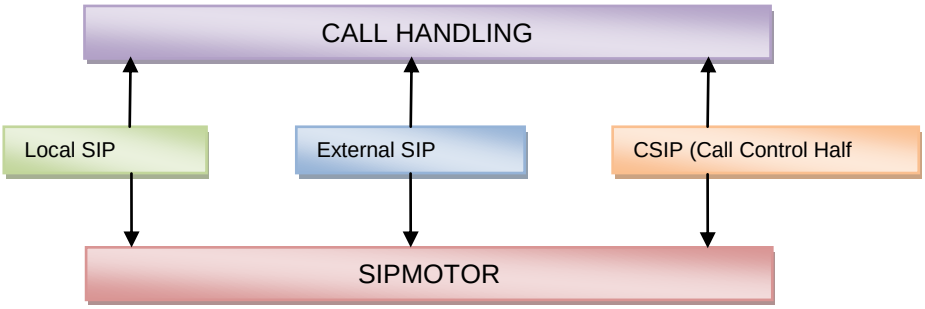
```
+---------+----------------------+------+----------+
| Number  |Name                  |Ext GW|Registered|
+---------+----------------------+------+----------+
| 60001   |       Dujardin Loulou|000001|        No|
+---------+----------------------+------+----------+
```

For each user directory number,the next information are present:

- the “Number” corresponds to the directory number of the SIP extension.
- the “Name” corresponds the name of the SIP extension.
- the “Ext GW” corresponds to the associated external SIP gateway linked to this SIP Device.
- the “Registered” gives the information to know if the SIP device is registered on OXE side.

11.6 Link between SIPMOTOR traces and Call Handling traces

11.6.1 Call Handling / SIPMOTOR links implementation



The local SIP gateway “link” is used for the local SIP elements

- The SIP devices
- The external SIP Voice Mail

The external SIP gateways “link” are used for the connection between an external SIP equipment to the OXE

- SIP carriers
- SIP applications (IVR, call center, etc...)

The Call Control Half Com “link” is used for the SIP extension users (SEPLOS), it corresponds to the “CSIP” function.

According to the declaration type of the SIP equipment on the OXE, the behavior will be different on the SIPMOTOR side, and also on the Call Handling side.

The exchanges between the SIPMOTOR to the Call Handling are different according to this declaration.

11.6.2 General view

When an issue appears in case of SIP equipment involved on the communication, it is important to check if the problem is from the SIPMOTOR or from the Call Handling.

It is important to make the 2 traces simultaneously in case of problem.

When a call is done, we can see on the motortrace the exchange between the SIPMOTOR to the Call handling.

- Exchange from Call Handling to SIPMOTOR on SIPMOTOR traces:

```
[display_ipc_in] ------------ Begin ---------------
.
.
.
[display_ipc_in] ------------- End ----------------
```

- Exchange from SIPMOTOR to Call Handling on SIPMOTOR traces:

```
[display_ipc_out] ------------ Begin ---------------
.
.
.
[display_ipc_out] ------------- End ----------------
```

- Exchange from Call Handling to SIPMOTOR on Call Handling traces:

```
+------------------------------------------------------------+
| Message sent UA (neqt : XXXX-0) ----> SIP
```

- Exchange from SIPMOTOR to Call Handling on Call Handling traces:

```
+------------------------------------------------------------+
| Message received SIP ----> UA (neqt : XXXX)
```

11.6.3 "neqt" link between SIPMOTOR and Call Handling traces

When traces are done on OXE to find the cause of the issue, it is important to link the call on the SIPMOTOR trace and the Call Hanling trace, for this check the "neqt" number used (the neqt is 2250 in the next examples)

- On SIPMOTOR traces:
  - For incoming call, the neqt is seen before the "display_ipc_out" message:

```
Mon May 28 14:22:38 2012 [CMotorCallManager::insertCallwithEqt] CMotorCall 2250 inserted.
Mon May 28 14:22:38 2012 11f7[sendLgEvtSipCreate] Event sent on eqt : 2250
Mon May 28 14:22:38 2012 [display_ipc_out] ------------ Begin ---------------
Mon May 28 14:22:38 2012 Id : -1
Mon May 28 14:22:38 2012 INVITE
```

  - For outgoing call, the neqt is given on the "display_ipc_in" message from the Call handling

```
Mon May 28 14:27:48 2012 [display_ipc_in] ------------ Begin ---------------
Mon May 28 14:27:48 2012 neqt : 2250 Id : -1
Mon May 28 14:27:48 2012 INVITE
```

- On Call Handling traces:
  - For incoming call, the neqt is seen with this message:

```
(215701:000005) SIP : message INVITE arrive sur le neqt : 2250.
(215701:000006) init_data_network
(215701:000007) init_data_network FIN
(215701:000008) SIP : ctrl_sip evt : 10752.
(215701:000009) +-------------------------------------------------------------+
(215701:000010) | Message received SIP ----> UA (neqt : 2250)
```

  - For outgoing call, the neqt is seen with this message:

```
(222651:000188) SIP : [send_to_motor] ipcSend resultat : 0 sur eqt : 2250
(222651:000189) SIP : [ipc_send] envoi du message : 10752.
(222651:000190) +-------------------------------------------------------------+
(222651:000191) | Message sent UA (neqt : 2250-0) ----> SIP
```

For traces analyses, follow all the exhanges using this neqt, it is not possible to get more than one active call using this "neqt". When the call is released, this "neqt" is freed for another call.

The "neqt" number can correspond to:

- A SIP extension, the same everytime.
- A time slot of the SIP Trunk Group used on the local SIP gateway for SIP device user, different according to which time slot is used.
- A time slot of the SIP Trunk Group used on the local SIP gateway for SIP external Voice Mail, different according to which time slot is used.
- A time slot of the SIP Trunk Group used for the external SIP gateway, different according to which time slot is used.

11.7 Information on the SIPMOTOR traces

On the SIPMOTOR traces, information are between “[...]”. These information are important to understand the information after it and to troubleshoot the issue.

Examples:

- [CCall::receiveRequest] INVITE: The SIPMOTOR has received a SIP request and the request is an INVITE.
- [CTransaction::changeState]: The SIPMOTOR has changed the state of a transaction.
- [getFromHeader]: the SIPMOTOR gets the information from the FROM header in case of SIP incoming call.
- [isDomainFromGwExt]: the SIPMOTOR checks if the information from the domain part of the FROM corresponds to an external SIP gateway.

The information “event” and “message” are in relation with the direction of the call and the SIP message:

- “event” is for the Call Handling.
- “message” is for the SIPMOTOR.

The information between the [...] are more or less understandable, they can help to find the root cause of the issue.

11.8 Follow a call on the SIPMOTOR trace

For SIP point of view, the call can be followed by the Call-ID, but on the SIPMOTOR, there are information for calls distinctions

- We have the “neqt” number, it is used to link the SIPMOTOR and Call Handling traces
- The Session reference is used to follow the call.
  - On this example, the Session reference is “1173”

```
Mon May 28 15:21:04 2012 1173[CMotorCall::sipUriType] sip Uri.
...
Mon May 28 15:21:04 2012 1173[CMotorCall::getUserType] seplos station crypto=0.
...
Mon May 28 15:21:04 2012 1173[CMotorCall::emitInviteMessage]  To: "Xlite PC" sip:31023@oxe-
ov.alcatel.fr;user=phone
...
Mon May 28 15:21:04 2012 1173[CMotorCall::inviteBuildContact]  Contact: sip:31004@oxe-ov.alcatel.fr
...
Mon May 28 15:21:04 2012 1173 [CCall::makeGenericRequest] INVITE
```

  - To find this Session reference for an outgoing call, search for “[CMotorCall::sipUriType] sip Uri.” before the INVITE sent to the remote SIP equipment.
  - To find this Session reference for an incoming call, search for “[CCall::receiveRequest] INVITE” after the INVITE received from the remote SIP equipment.

- The transation reference, this value can be used to follow the transaction status evolution and to get information about this transaction

  - On this example, the transaction reference is “21be”

```
Mon May 28 15:21:04 2012 21be [CTransaction::changeState] STATE CHANGED TO INITIAL
...
Mon May 28 15:21:04 2012 21be [CTransaction::changeState] STATE CHANGED TO CALLING
...
Mon May 28 15:21:04 2012 21be [CTransaction::startTimer] Timer A is started (delay = 500 ms)
Mon May 28 15:21:04 2012 21be [CTransaction::startTimer] Timer B is started (delay = 4000 ms)
...
Mon May 28 15:21:04 2012 21be [CTransaction::changeState] STATE CHANGED TO PROCEEDING
...
Mon May 28 15:21:08 2012 21be [CTransaction::changeState] STATE CHANGED TO TERMINATED
```

  - To find this transaction reference for an outgoing call, search for “STATE CHANGED TO INITIAL” before the INVITE sent to the remote SIP equipment.

  - To find this transaction reference for an incoming call, search for “STATE CHANGED TO INITIAL” after the INVITE received from the remote SIP equipment.

  - For one transaction, there is a pair of reference, a “clone” reference associated to the main one, if the main one is 21be, the second reference is 21bf associated with the 200ok receive or sent. This reference is seen with this message after the 200ok.

```
Mon May 28 15:21:08 2012 21bf [CTransaction::CTransaction] Transaction is cloned in 4 state
```

- The dialog reference, this value can be used to follow the dialog evolution and to get information about this dialog
  - On this example, the dialog reference is “158a”

```
Mon May 28 15:21:04 2012 158a [CDialog::createRequest]
Mon May 28 15:21:04 2012 158a [CDialog::buildServicesForAllRequest]
Mon May 28 15:21:04 2012 158a [CDialog::createInviteRequest]
...
Mon May 28 15:21:04 2012 158a [CDialog::onTransactionState(pTrans = 21be, previousState = Terminated, currentState
= Initial, reason = None]
...
Mon May 28 15:21:08 2012 158a [CDialog::receiveResponse]
Mon May 28 15:21:08 2012 158a [CDialog::receiveResponse] create a CONFIRMED dialog
```

  - To find this dialog reference for an outgoing call, search for “CDialog::createRequest” before the INVITE sent to the remote SIP equipment.

  - To find this dialog reference for an incoming call, search for “CDialog::receiveRequest” after the INVITE received from the remote SIP equipment.

  - For one dialog, there is a pair of reference, a “clone” reference associated to the main one, if the main one is 158a, the second reference is 158b associated with the 200ok receive or sent. This reference is seen with this message after the 200ok.

```
Mon May 28 15:21:08 2012 158b [CDialog::CDialog] look for the transaction #0, transaction key =
z9hG4bKca60f1097ab026913ca3bf56995162be
```

- This Information links the transaction to the dialog.

```
Mon May 28 15:21:04 2012 158a [CDialog::onTransactionState(pTrans = 21be, previousState = Terminated, currentState
= Initial, reason = None]
```

- For the dialog, the transaction reference is linked. The dialog “158a” is linked to the transaction “21be”.

- There is the same link for the “clone” references.

```
Mon May 28 15:21:08 2012 158b [CDialog::onTransactionState(pTrans = 21bf, previousState = Proceeding, currentState
= Completed, reason = Final resp reception]
```

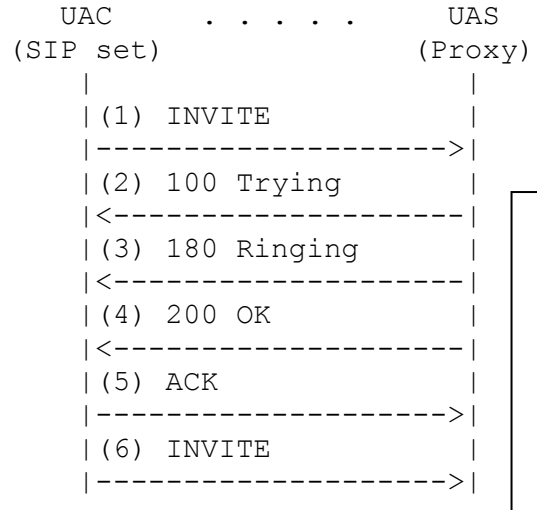
The SIPMOTOR is using references for INVITE treatment:

- The Session reference, this one is unique for the complete call (from INVITE to the 200ok of the BYE)

- The Dialog references, 2 references are used:
  - The main one is created when the INVITE is sent or received
  - The clone one, used to change the dialog state according to the transactions used for a new event on the call (put on hold, transfer, etc...)

- The Transaction references, the number of references depends of the call events (put on hold, transfer, etc...)
  - The main one is created when the INVITE is sent or received
  - The other ones are created if an event is coming for the dialog associated (ACK, BYE, REINVITE, REFER, etc...)

A permanent link is done between the Dialog (main and clone) to the Transactions (main and clones), here an example for an incoming call with 2 REINVITEs and a BYE at the end:



(1) Assignation a reference to the session, dialog and transaction

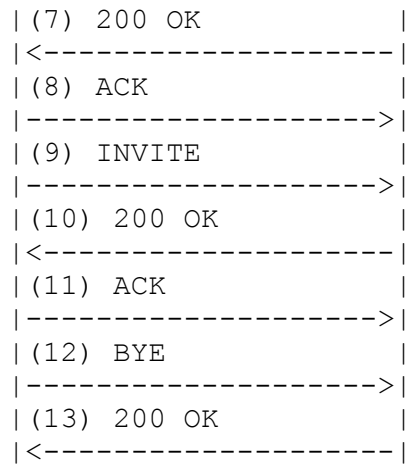
(4) Creation of the clone dialog and the first clone transaction, associated to the clone dialog

(5) First clone transaction terminated

(6) Creation of the second clone transaction for the first REINVITE, associated to the clone dialog

(8) Second clone transaction terminated

(9) Creation of the third clone transaction for the second REINIVTE, associated to the clone dialog



11.9 Traces analyses

11.9.1 Incoming SIP call using a SIP Trunk Group: SIPMOTOR point of view

Here an example of incoming call from a SIP device to an IPtouch.

```
Mon May 28 16:41:57 2012 RECEIVE MESSAGE FROM NETWORK (135.118.226.39:25648 [UDP])
----------------------utf8-----------------------
INVITE sip:31004@oxe-ov.alcatel.fr SIP/2.0
Via: SIP/2.0/UDP 135.118.226.39:25648;branch=z9hG4bK-d87543-46534e582323f252-1--d87543-;rport
Max-Forwards: 70
Contact: <sip:31024@135.118.226.39:25648>
To: "31004"<sip:31004@oxe-ov.alcatel.fr>
From: "PC_sip_device"<sip:31024@oxe-ov.alcatel.fr>;tag=f6448c0c
Call-ID: ZWEwMGI4YjUxNjMyOWRlZmEyNWEzYThmNzI4NDUzMGM.
CSeq: 1 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO
Content-Type: application/sdp
User-Agent: Sip Phone
Content-Length: 315

v=0
o=- 3 2 IN IP4 135.118.226.39
s=Sip_Phone
c=IN IP4 135.118.226.39
t=0 0
m=audio 7888 RTP/AVP 8 18 101
a=fmtp:18 annexb=no
a=fmtp:101 0-15
a=rtpmap:101 telephone-event/8000
a=sendrecv
a=x-rtp-session-id:A56A9738C0BC4CEF8087E10840231621
------------------------------------------------
```

The information “RECEIVE MESSAGE FROM NETWORK (135.118.226.39:25648 [UDP])” is important to know that the call is an incoming one from the SIP equipment 135.118.226.39 in UDP.

- The SIPMOTOR checks the Call-Id to know if this INVITE is an INVITE or a REINVITE.

```
Mon May 28 16:41:57 2012 1153 [CCall::getDialog] Confirmed Dialog is not found (ID = ;f6448c0c)
Mon May 28 16:41:57 2012 1153 [CCall::getDialog] Initial Dialog Server not found
```

Here, it is an INVITE, because the dialog is not found.

- The transaction and the dialog are put in place.

```
Mon May 28 16:41:57 2012 21a5 [CTransaction::changeState] STATE CHANGED TO INITIAL
...
Mon May 28 16:41:57 2012 156c [CDialog::onTransactionState(pTrans = 21a5, previousState = Terminated, currentState
= Initial, reason = None]
```

Here, the transaction reference is “21a5” and the dialog reference is “156c”.

- The transaction status is changed, because the dialog is initiated.

```
Mon May 28 16:41:57 2012 21a5 [CTransaction::changeState] STATE CHANGED TO PROCEEDING
Mon May 28 16:41:57 2012 21a5 [CTransaction::changeState] notifying the parent dialog
```

When a transaction is linked to a dialog, the transaction changed from INITIAL to PROCEEDING.

- The SIPMOTOR generates the 100 Trying.

```
Mon May 28 16:41:57 2012 SEND MESSAGE TO NETWORK (135.118.226.39:25648 [UDP]) (BUFF LEN = 346)
----------------------utf8-----------------------
SIP/2.0 100 Trying
To: "31004" <sip:31004@oxe-ov.alcatel.fr>
From: "PC_sip_device" <sip:31024@oxe-ov.alcatel.fr>;tag=f6448c0c
Call-ID: ZWEwMGI4YjUxNjMyOWRlZmEyNWEzYThmNzI4NDUzMGM.
CSeq: 1 INVITE
Via: SIP/2.0/UDP 135.118.226.39:25648;received=135.118.226.39;branch=z9hG4bK-d87543-46534e582323f252-1--d87543-
;rport=25648
Content-Length: 0
-------------------------------------------------
```

- The SIPMOTOR checks the Session Timer for the call.

```
Mon May 28 16:41:57 2012 [CSessionTimerContext::CSessionTimerContext] New CSessionTimerContext from request
(Server, UA)
Mon May 28 16:41:57 2012 [CSessionTimerContext::updateAfterRefreshReception] Update CSessionTimerContext (refresh
reception)
Mon May 28 16:41:57 2012 [CSessionTimerContext::updateSessionExpires] Session-Expires updated : 0
Mon May 28 16:41:57 2012 [CSessionTimerContext::setRefreshMethod] Allow refreshMethod=INVITE
```

In this case, the SIP equipment doesn’t send “Session timer” information because the value is 0 (updated : 0).

- The SIPMOTOR makes the link between the dialog, transaction, the branch and the Cseq number.

```
Mon May 28 16:41:57 2012 156c [CDialog::addTransaction] added transaction 21a5 with branch z9hG4bK-d87543-
46534e582323f252-1--d87543-, with CSeq 1
```

The “branch” is a parameter added to the “via” to identify it. Regarding rfc3261, all the branch values must start by “z9hG4bK”.

The CSeq is used to indentify and to order a transaction, it consists of a sequence number and a method.

- The SIPMOTOR checks for which OXE equipment the call is from.

```
Mon May 28 16:41:57 2012 [isDomainFromGwExt] Host from request is : 172.27.142.53.
Mon May 28 16:41:57 2012 [isDomainFromGwExt] User from request is : 31024
Mon May 28 16:41:57 2012 [domain not from an External Gateway.
Mon May 28 16:41:57 2012 1153[CMotorCall::setFilterUsedMode] To be traced = 0
Mon May 28 16:41:57 2012 1153[CMotorCall::initOfUserType] values are reseted
Mon May 28 16:41:57 2012 [getFromHeader] displayName="PC_sip_device".
Mon May 28 16:41:57 2012 [getFromHeader] =31024@oxe-ov.alcatel.fr.
Mon May 28 16:41:57 2012 [getFromHeader] clirPresent=0.
Mon May 28 16:41:57 2012 [isAddrInDico] user=31024 host=oxe-ov.alcatel.fr
Mon May 28 16:41:57 2012 [isUserInDico] 31024@oxe-ov.alcatel.fr
Mon May 28 16:41:57 2012 [isUserInDico]  found in the dictionnary.
Mon May 28 16:41:57 2012 [isAddrInDico] sip device station OK
```

- The SIPMOTOR checks first if the domain part from the PAI, and of the FROM if no PAI, to see if the call is for an external SIP gateway.
- Here, we can see that the call is from a SIP Device.

- The SIPMOTOR checks for who the call is done .

```
Mon May 28 16:41:57 2012 [isAddrInDico] user=31004 host=oxe-ov.alcatel.fr
Mon May 28 16:41:57 2012 [isUserInDico] 31004@oxe-ov.alcatel.fr
Mon May 28 16:41:57 2012 isUserInDico] NOT found in the dictionnary.
Mon May 28 16:41:57 2012 [isAddrInDico] other sip user
```

Here the call is for an “other sip user”, that means the call is for a non SIP user, corresponding to a legacy set (IPtouch).

- The SIPMOTOR checks the number of licenses available.

```
Mon May 28 16:41:57 2012 1153[CMotorCall::methodInviteReceived] nb available licenses=25
```

Here the number of licenses is 25, that means, 25 calls are possible on SIP using a SIP Trunk Group or SEPLOS users.

- The SIPMOTOR checks if the IP address recieved is managed on an IP domain.

```
Mon May 28 16:41:57 2012    The recevied host 135.118.226.39
Mon May 28 16:41:57 2012  Trying to find the ip address in domain list
Mon May 28 16:41:57 2012  The entry dom : 141 add_type=1
Mon May 28 16:41:57 2012  The entry dom ip low :172.27.141.165
Mon May 28 16:41:57 2012  The entry ipaddress from low :135.118.226.39
Mon May 28 16:41:57 2012  The entry compare :1
Mon May 28 16:41:57 2012  The entry compare 2 :0
Mon May 28 16:41:57 2012  iplink_is_good_range_for_reg
...
Mon May 28 16:41:57 2012  The user domain is  142
```

Here, the IP address of the SIP equipment corresponds to the IP domain 142.

If the IP address doesn’t match an IP domain, the SIPMOTOR returns:

```
Mon May 28 16:41:57 2012  The user is ipadd  not in any Domain range return state as -1
```

- The SIPMOTOR checks the SDP received on the INVITE.

```
Mon May 28 16:41:57 2012 [checkSdpValidity] Media 0 type 1 contains 3 formats.
Mon May 28 16:41:57 2012 [checkSdpValidity] Format : 8.
Mon May 28 16:41:57 2012 1153[CMotorCall::isCryptoAuthorized] user crypto=0.
Mon May 28 16:41:57 2012 [convertSdpIntoTsdp] No Direction in the session part.
Mon May 28 16:41:57 2012 [convertSdpIntoTsdp] Check the direction in Session part - result:0.
Mon May 28 16:41:57 2012 [convertSdpIntoTsdp] media AUDIO detected (previous crypto=0).
Mon May 28 16:41:57 2012 [convertAudioMedia] The audio media contains  3 format(s).
Mon May 28 16:41:57 2012 [convertAudioMedia] Format 0 is 8.
Mon May 28 16:41:57 2012 [convertAudioMedia] Format 1 is 18.
Mon May 28 16:41:57 2012 [convertAudioMedia] Format 2 is 101.
Mon May 28 16:41:57 2012 [convertAudioMedia] 101.
Mon May 28 16:41:57 2012 [convertAudioMedia] Format is DTMF:101.
Mon May 28 16:41:57 2012 [convertAudioMedia] Direction is sendrecv.
Mon May 28 16:41:57 2012 [convertAudioMedia] Connection address retrieved in sdp: 135.118.226.39.
Mon May 28 16:41:57 2012 [convertIPStrIntoTuipv] 135.118.226.39 => 135.118.226.39
Mon May 28 16:41:57 2012 [display_sdp] address =135.118.226.39
Mon May 28 16:41:57 2012 [display_sdp] direction=0.
Mon May 28 16:41:57 2012 [convertSdpIntoTsdp] only one media taken into account xxx crypto_index=0 clear media=1
Mon May 28 16:41:57 2012 [convertSdpIntoTsdp] crypto_index=0 clear media=1.
```

The SDP contains in this SDP three formats of medias (8, 18 and 101), the direction is “sendrecv” meaning in both direction and the IP address of connection is 135.118.226.39.

- The message to Call Handling is prepared and sent to it.

```
Mon May 28 16:41:57 2012 1153[sendLgEvtSipCreate] Event sent on eqt : 2250
Mon May 28 16:41:57 2012 [display_ipc_out] ------------ Begin ---------------
Mon May 28 16:41:57 2012 Id : -1
Mon May 28 16:41:57 2012 INVITE
Mon May 28 16:41:57 2012 REQUEST URI : <> 31004@oxe-ov.alcatel.fr:5060 ; user=name
Mon May 28 16:41:57 2012 FROM : <PC_sip_device> 31024@oxe-ov.alcatel.fr:5060 ; user=name
Mon May 28 16:41:57 2012 TO : <"31004"> 31004@oxe-ov.alcatel.fr:5060 ; user=name
Mon May 28 16:41:57 2012 CAC : 0
Mon May 28 16:41:57 2012   CAC ADDRESS :
Mon May 28 16:41:57 2012 CAC-CSBU info : UNKNOWN
Mon May 28 16:41:57 2012 CLIR : 0
Mon May 28 16:41:57 2012 Prack Required : 0
Mon May 28 16:41:57 2012 Allow Update : 0
Mon May 28 16:41:57 2012 SDP :
Mon May 28 16:41:57 2012   ADDRESS : 135.118.226.39 :7888
Mon May 28 16:41:57 2012   ALGOS :
Mon May 28 16:41:57 2012     PCMA
Mon May 28 16:41:57 2012     G729
Mon May 28 16:41:57 2012     101
Mon May 28 16:41:57 2012   DIRECTION : SEND & RECEIVE
Mon May 28 16:41:57 2012   crypto index : 0
Mon May 28 16:41:57 2012    N_GW_EXT : -1
Mon May 28 16:41:57 2012 [display_ipc_out] ------------- End ----------------
```

The call is sent to the Call handling on neqt 2250, regarding the type of SIP equipment detected by the SIPMOTOR, some information are added or not on this message.

- All the information about this call are sent to the Stand-By CPU.

```
Mon May 28 16:41:57 2012 SendToSipgwCpuSec: Message sent to the STAND-BY CPU
Mon May 28 16:41:57 2012 [receiveInviteMessage] send RemoteSdp to the StandBy.
Mon May 28 16:41:57 2012 SendToSipgwCpuSec: Message sent to the STAND-BY CPU
```

The information are sent to the Stand-By, like this, in case of bascul the SIP call will not be lost and known on the new main CPU

- The Call handling sends back an answer for this INVITE.

```
Mon May 28 16:41:57 2012 [display_ipc_in] ------------ Begin ---------------
Mon May 28 16:41:57 2012 neqt : 2250 Id : -1
Mon May 28 16:41:57 2012 INFORMATIONAL
Mon May 28 16:41:57 2012 xx :       80
Mon May 28 16:41:57 2012    RELATIVE REQUEST : INVITE
Mon May 28 16:41:57 2012 [display_ipc_in] ------------- End ----------------
```

A “180 Ringing” is sent to the SIPMOTOR without SDP

- The Call handling sends back an answer for this INVITE.

```
Mon May 28 16:41:57 2012 SEND MESSAGE TO NETWORK (135.118.226.39:25648 [UDP]) (BUFF LEN = 547)
----------------------utf8-----------------------
SIP/2.0 180 Ringing
Allow: INVITE, ACK, CANCEL, BYE, PRACK, NOTIFY, REFER, SUBSCRIBE, OPTIONS, UPDATE
Contact: sip:oxe-ov.alcatel.fr
User-Agent: OmniPCX Enterprise R10.0 j1.410.45
To: "31004" <sip:31004@oxe-ov.alcatel.fr>;tag=15654dedb5658c165fbba7b0026e6ae9
From: "PC_sip_device" <sip:31024@oxe-ov.alcatel.fr>;tag=f6448c0c
Call-ID: ZWEwMGI4YjUxNjMyOWRlZmEyNWEzYThmNzI4NDUzMGM.
CSeq: 1 INVITE
Via: SIP/2.0/UDP 135.118.226.39:25648;received=135.118.226.39;branch=z9hG4bK-d87543-46534e582323f252-1--d87543-
;rport=25648
Content-Length: 0
-------------------------------------------------
```

A “180 Ringing” is sent to the SIPMOTOR without SDP

- For each SIP call event, a message is send to the Stand-By CPU.

```
Mon May 28 16:41:57 2012 [receiveInformationalEvent] UpdateContext send on the StandBy.
```

- The Call handling sends a new answer for this INVITE.

```
Mon May 28 16:41:58 2012 [display_ipc_in] ------------ Begin ---------------
Mon May 28 16:41:58 2012 neqt : 2250 Id : -1
Mon May 28 16:41:58 2012 SUCCESSFUL
Mon May 28 16:41:58 2012 xx :        0
Mon May 28 16:41:58 2012    RELATIVE REQUEST : INVITE
Mon May 28 16:41:58 2012 CLIR : 0
Mon May 28 16:41:58 2012 COLP : 1
Mon May 28 16:41:58 2012 CAC-CSBU info : UNKNOWN
Mon May 28 16:41:58 2012 SDP :
Mon May 28 16:41:58 2012   ADDRESS : 172.27.142.64 :32514
Mon May 28 16:41:58 2012   ALGOS :
Mon May 28 16:41:58 2012     G729
Mon May 28 16:41:58 2012     101
Mon May 28 16:41:58 2012   DIRECTION : SEND & RECEIVE
Mon May 28 16:41:58 2012   crypto index : 0
Mon May 28 16:41:58 2012 [display_ipc_in] ------------- End ----------------
```

A “200 ok” is sent to the SIPMOTOR with SDP

- The SIPMOTOR checks if the SDP given is compatible with the SDP received in the INVITE.

```
Mon May 28 16:41:58 2012 1153[CMotorCall::makeResponseSdp] Audio media.
Mon May 28 16:41:58 2012 1153[CMotorCall::appendAudioAttributToMedia] Direction: 0.
Mon May 28 16:41:58 2012 1153[CMotorCall::appendAudioAttributToMedia] format 101
Mon May 28 16:41:58 2012 1153[CMotorCall::makeResponseSdp] fromSdp.getMediaDesciprionCount :1
Mon May 28 16:41:58 2012 [sameCodec] accepted Format : 18.
Mon May 28 16:41:58 2012 [sameCodec] requested Format : 8.
Mon May 28 16:41:58 2012 [sameCodec] requested Format : 18.
Mon May 28 16:41:58 2012 [sameCodec] same Format.
Mon May 28 16:41:58 2012 1153[CMotorCall::mediaAccepted] Media accepted: m=audio 32514 RTP/AVP 18 101
a=sendrecv
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=ptime:20
a=maxptime:40
a=rtpmap:101 telephone-event/8000
.
```

The codecs from the INVITE were 8 and 18, on the answer we have 18, in that case the call is accepted by SIPMOTOR for SDP point of view.

- The SIPMOTOR is changing the status of the dialog.

```
Mon May 28 16:41:58 2012 156c [CDialog::createResponse] create a CONFIRMED dialog
```

Due to this, the dialog reference and transaction reference are changed (internal SIPMOTOR functionning).

```
Mon May 28 16:41:58 2012 156d [CDialog::CDialog] look for the transaction #0, transaction key = z9hG4bK-d87543-
46534e582323f252-1--d87543-
Mon May 28 16:41:58 2012 156d [CDialog::CDialog] copy the transaction #0, transaction key = z9hG4bK-d87543-
46534e582323f252-1--d87543-
Mon May 28 16:41:58 2012 21a6 [CTransaction::CTransaction] Transaction is cloned in 4 state
```

The dialog reference is changed form “156c” to “156d”.
The transaction reference is changed from “21a5” to “21a6”.

- The SIPMOTOR is changing the status of the dialog.

```
1338216118 -> Mon May 28 16:41:58 2012 SEND MESSAGE TO NETWORK (135.118.226.39:25648 [UDP]) (BUFF LEN = 974)
----------------------utf8-----------------------
SIP/2.0 200 OK
Allow: INVITE, ACK, CANCEL, BYE, PRACK, NOTIFY, REFER, SUBSCRIBE, OPTIONS, UPDATE
Contact: sip:oxe-ov.alcatel.fr
Supported: replaces,timer,path,100rel
User-Agent: OmniPCX Enterprise R10.0 j1.410.45
Session-Expires: 1800;refresher=uas
P-Asserted-Identity: "IPtouch 172.27.1" <sip:31004@oxe-ov.alcatel.fr;user=phone>
Content-Type: application/sdp
To: "31004" <sip:31004@oxe-ov.alcatel.fr>;tag=15654dedb5658c165fbba7b0026e6ae9
From: "PC_sip_device" <sip:31024@oxe-ov.alcatel.fr>;tag=f6448c0c
```

The 200ok sent to the remote SIP equipment is generated with information from the INVITE received and from the 200ok answer from the Call Handling.

- The SIPMOTOR is changing the status of the transaction.

```
Mon May 28 16:41:58 2012 21a6 [CTransaction::changeState] STATE CHANGED TO COMPLETED
```

- The retransmission timers are started.

```
Mon May 28 16:41:58 2012 21a6 [CTransaction::startTimer] Timer G is started (delay = 500 ms)
Mon May 28 16:41:58 2012 21a6 [CTransaction::startTimer] Timer H is started (delay = 32000 ms)
```

- The SIPMOTOR receives a ACK for the 200ok.

```
Mon May 28 16:41:59 2012 RECEIVE MESSAGE FROM NETWORK (135.118.226.39:25648 [UDP])
----------------------utf8-----------------------
ACK sip:oxe-ov.alcatel.fr SIP/2.0
Via: SIP/2.0/UDP 135.118.226.39:25648;branch=z9hG4bK-d87543-b00f692e5d3a246e-1--d87543-;rport
Max-Forwards: 70
Contact: <sip:31024@135.118.226.39:25648>
To: "31004"<sip:31004@oxe-ov.alcatel.fr>;tag=15654dedb5658c165fbba7b0026e6ae9
From: "PC_sip_device"<sip:31024@oxe-ov.alcatel.fr>;tag=f6448c0c
Call-ID: ZWEwMGI4YjUxNjMyOWRlZmEyNWEzYThmNzI4NDUzMGM.
CSeq: 1 ACK
User-Agent: Sip Phone
Content-Length: 0
-------------------------------------------------
```

- The SIPMOTOR is changing the status of the transaction.

```
Mon May 28 16:41:59 2012 21a6 [CTransaction::changeState] STATE CHANGED TO TERMINATED
```

- The retransmission timers are freed.

```
Mon May 28 16:41:59 2012 21a6 [CTransaction::freeTimerToken] Timer G is freed
Mon May 28 16:41:59 2012 21a6 [CTransaction::freeTimerToken] Timer H is freed
```

- The SIPMOTOR is changing the status of the dialog.

```
Mon May 28 16:41:59 2012 156d [CDialog::receiveAckRequest] the INVITE request is terminated
```

- The ACK is sent to the Call Handling.

```
Mon May 28 16:41:59 2012 [display_ipc_out] ------------ Begin ---------------
Mon May 28 16:41:59 2012 Id : -1
Mon May 28 16:41:59 2012 ACK
Mon May 28 16:41:59 2012 [display_ipc_out] ------------- End ----------------
```

After call establishment, the call can be released by the OXE or by the remote SIP equipment.

- Call released by the Call Handling:

  - The BYE is sent from the Call Handling.

```
Mon May 28 16:42:00 2012 [display_ipc_in] ------------ Begin ---------------
Mon May 28 16:42:00 2012 neqt : 2250 Id : -1
Mon May 28 16:42:00 2012 BYE
Mon May 28 16:42:00 2012 [display_ipc_in] ------------- End ----------------
```

  - Creation of a new transaction for the BYE.

```
Mon May 28 16:42:00 2012 21a7 [CTransaction::changeState] STATE CHANGED TO INITIAL
```

The BYE is a new transaction for a SIP call, in that case, the transaction reference it is “21a7”, and the status is “INITIAL”.

  - The BYE is sent to the remote SIP equipment.

```
Mon May 28 16:42:00 2012 SEND MESSAGE TO NETWORK (135.118.226.39:25648 [UDP]) (BUFF LEN = 454)
----------------------utf8-----------------------
BYE sip:31024@135.118.226.39:25648 SIP/2.0
Supported: replaces,timer,path,100rel
User-Agent: OmniPCX Enterprise R10.0 j1.410.45
To: sip:31024@oxe-ov.alcatel.fr;tag=f6448c0c
From: "31004" <sip:31004@oxe-ov.alcatel.fr>;tag=15654dedb5658c165fbba7b0026e6ae9
Call-ID: ZWEwMGI4YjUxNjMyOWRlZmEyNWEzYThmNzI4NDUzMGM.
CSeq: 1948273321 BYE
Via: SIP/2.0/UDP 172.27.142.53;branch=z9hG4bK9f0b6b39121b23d361a5f6a8101aaa90
Max-Forwards: 70
Content-Length: 0
-------------------------------------------------
```

  - The SIPMOTOR changes the transaction state.

```
Mon May 28 16:42:00 2012 21a7 [CTransaction::changeState] STATE CHANGED TO TRYING
```

  - The retransmission timers are started.

```
Mon May 28 16:42:00 2012 21a7 [CTransaction::startTimer] Timer E is started (delay = 500 ms)
Mon May 28 16:42:00 2012 21a7 [CTransaction::startTimer] Timer F is started (delay = 16000 ms)
```

  - The 200ok of the BYE request is received from the remote SIP equipment.

```
Mon May 28 16:42:00 2012 RECEIVE MESSAGE FROM NETWORK (135.118.226.39:25648 [UDP])
----------------------utf8-----------------------
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.27.142.53;branch=z9hG4bK9f0b6b39121b23d361a5f6a8101aaa90
Contact: <sip:31024@135.118.226.39:25648>
To: <sip:31024@oxe-ov.alcatel.fr>;tag=f6448c0c
From: "31004"<sip:31004@oxe-ov.alcatel.fr>;tag=15654dedb5658c165fbba7b0026e6ae9
Call-ID: ZWEwMGI4YjUxNjMyOWRlZmEyNWEzYThmNzI4NDUzMGM.
CSeq: 1948273321 BYE
User-Agent: Sip Phone
Content-Length: 0
-------------------------------------------------
```

- The SIPMOTOR changes this transaction state.

```
Mon May 28 16:42:00 2012 21a7 [CTransaction::changeState] STATE CHANGED TO COMPLETED
```

- The retransmission timers are freed.

```
Mon May 28 16:42:00 2012 21a7 [CTransaction::freeTimerToken] Timer E is freed
Mon May 28 16:42:00 2012 21a7 [CTransaction::freeTimerToken] Timer F is freed
```

- The 200ok of the BYE request is sent to the Call Handling.

```
Mon May 28 16:42:00 2012 [display_ipc_out] ------------ Begin ---------------
Mon May 28 16:42:00 2012 Id : -1
Mon May 28 16:42:00 2012 SUCCESSFUL
Mon May 28 16:42:00 2012 xx :        0
Mon May 28 16:42:00 2012    RELATIVE REQUEST : BYE
Mon May 28 16:42:00 2012 CAC-CSBU info : UNKNOWN
Mon May 28 16:42:00 2012 CLIR : 0
Mon May 28 16:42:00 2012 COLP : 0
Mon May 28 16:42:00 2012 [display_ipc_out] ------------- End ----------------
```

- The Call Handling sent a message to the SIPMOTOR to release the “neqt” associated to this SIP call

```
Mon May 28 16:42:00 2012 [display_ipc_in] ------------ Begin ---------------
Mon May 28 16:42:00 2012 neqt : 2250 Id : -1
Mon May 28 16:42:00 2012 SIP EQT RELEASED
Mon May 28 16:42:00 2012 [display_ipc_in] ------------- End ----------------
```

- The SIPMOTOR acknowledge the release of the “neqt”

```
Mon May 28 16:42:00 2012 [display_ipc_out] ------------ Begin ---------------
Mon May 28 16:42:00 2012 Id : -1
Mon May 28 16:42:00 2012 SIP_EQT_RELEASE_ACK
Mon May 28 16:42:00 2012 [display_ipc_out] ------------- End ----------------
```

- The SIPMOTOR kills the SIP call

```
Mon May 28 16:42:00 2012 [CMotorCallManager::onIncomingEvent] killSession.
Mon May 28 16:42:00 2012 1153 [CCall::killSession]
```

- The SIPMOTOR changes the state of the transactions

```
Mon May 28 16:42:00 2012 21a5 [CTransaction::changeState] STATE CHANGED TO TERMINATED
...
Mon May 28 16:42:00 2012 21a7 [CTransaction::changeState] STATE CHANGED TO TERMINATED
```

- Call released by the remote SIP equipment:

  - The BYE is received from the remote SIP equipment.

```
Mon May 28 16:42:00 2012 RECEIVE MESSAGE FROM NETWORK (135.118.226.39:25648 [UDP])
----------------------utf8----------------------
BYE sip:oxe-ov.alcatel.fr SIP/2.0
Via: SIP/2.0/UDP 135.118.226.39:25648;branch=z9hG4bK-d87543-cf501c2f3311d050-1--d87543-;rport
Max-Forwards: 70
Contact: <sip:31024@135.118.226.39:25648>
To: "31004"<sip:31004@oxe-ov.alcatel.fr>;tag=ba904e80f620e0f32593273ec97e818d
From: "PC_sip_device"<sip:31024@oxe-ov.alcatel.fr>;tag=b05ced13
Call-ID: NTEwZjI0M2VjZGY1YzExZTMzZWVjOGY2YzM0MmI5ODU.
CSeq: 2 BYE
User-Agent: Sip Phone
Content-Length: 0
------------------------------------------------
```

  - The SIPMOTOR checks if the dialog is already exist.

```
Mon May 28 16:42:00 2012 1153 [CCall::getDialog] Confirmed Dialog found
```

  - Creation of a new transaction for the BYE.

```
Mon May 28 16:42:00 2012 21a7 [CTransaction::changeState] STATE CHANGED TO INITIAL
```

The BYE is a new transaction for a SIP call, in that case, the transaction reference it is “21a7”, and the status is “INITIAL”.

  - The SIPMOTOR changes the transaction state.

```
Mon May 28 16:42:00 2012 21a7 [CTransaction::changeState] STATE CHANGED TO TRYING
```

  - The BYE is sent to the Call handling.

```
Mon May 28 16:42:00 2012 [display_ipc_out] ------------ Begin ---------------
Mon May 28 16:42:00 2012 Id : -1
Mon May 28 16:42:00 2012 BYE
Mon May 28 16:42:00 2012 [display_ipc_out] ------------- End -----------------
```

  - The Call Handling answers to the SIPMOTOR.

```
Mon May 28 16:42:00 2012 [display_ipc_in] ------------ Begin ---------------
Mon May 28 16:42:00 2012 neqt : 2250 Id : -1
Mon May 28 16:42:00 2012 SUCCESSFUL
Mon May 28 16:42:00 2012 xx :        0
Mon May 28 16:42:00 2012    RELATIVE REQUEST : BYE
Mon May 28 16:42:00 2012 CLIR : 0
Mon May 28 16:42:00 2012 COLP : 0
Mon May 28 16:42:00 2012 CAC-CSBU info : UNKNOWN
Mon May 28 16:42:00 2012 [display_ipc_in] ------------- End ---------------
```

  - The SIPMOTOR sends the 200 ok of the BYE to the remote SIP equipment.

```
Tue May 29 14:21:53 2012 SEND MESSAGE TO NETWORK (135.118.226.39:25648 [UDP]) (BUFF LEN = 546)
----------------------utf8----------------------
SIP/2.0 200 OK
Allow: INVITE, ACK, CANCEL, BYE, PRACK, NOTIFY, REFER, SUBSCRIBE, OPTIONS, UPDATE
Supported: replaces,timer,path,100rel
User-Agent: OmniPCX Enterprise R10.0 j1.410.45
To: "31004" <sip:31004@oxe-ov.alcatel.fr>;tag=ba904e80f620e0f32593273ec97e818d
From: "PC_sip_device" <sip:31024@oxe-ov.alcatel.fr>;tag=b05ced13
Call-ID: NTEwZjI0M2VjZGY1YzExZTMzZWVjOGY2YzM0MmI5ODU.
CSeq: 2 BYE
Via: SIP/2.0/UDP 135.118.226.39:25648;received=135.118.226.39;branch=z9hG4bK-d87543-cf501c2f3311d050-1--d87543-
;rport=25648
Content-Length: 0
```

- The SIPMOTOR changes the transaction state.

```
Mon May 28 16:42:00 2012 21a7 [CTransaction::changeState] STATE CHANGED TO COMPLETED
```

- The Call Handling sends a message to the SIPMOTOR to release the “neqt” associated to this SIP call

```
Mon May 28 16:42:00 2012 [display_ipc_in] ------------ Begin ---------------
Mon May 28 16:42:00 2012 neqt : 2250 Id : -1
Mon May 28 16:42:00 2012 SIP EQT RELEASED
Mon May 28 16:42:00 2012 [display_ipc_in] ------------- End ----------------
```

- The SIPMOTOR acknowledge the release of the “neqt”

```
Mon May 28 16:42:00 2012 [display_ipc_out] ------------ Begin ---------------
Mon May 28 16:42:00 2012 Id : -1
Mon May 28 16:42:00 2012 SIP_EQT_RELEASE_ACK
Mon May 28 16:42:00 2012 [display_ipc_out] ------------- End ----------------
```

- The SIPMOTOR kills the SIP call

```
Mon May 28 16:42:00 2012 [CMotorCallManager::onIncomingEvent] killSession.
Mon May 28 16:42:00 2012 1153 [CCall::killSession]
```

- The SIPMOTOR change the state of the transactions

```
Mon May 28 16:42:00 2012 21a5 [CTransaction::changeState] STATE CHANGED TO TERMINATED
...
Mon May 28 16:42:00 2012 21a7 [CTransaction::changeState] STATE CHANGED TO TERMINATED
```

11.9.2 Incoming SIP call using a SIP Trunk Group: Call Handling point of view

Here an example of incoming call from a SIP device to an IPtouch.

Traces option used :

>tuner km
>tuner clear-traces
>trc i
>actdbg all=off
>tuner +cpu +cpl +at hybrid=on
>actdbg sip=on abcf=on
>mtracer -a

- The call arrives on the SIPMOTOR, and sending to the Call Handling

```
(292779:000028) +------------------------------------------------------------+
(292779:000029) | Message received SIP ----> UA (neqt : 2250)
(292779:000030) | INVITE : 31004@oxe-ov.alcatel.fr:5060 ; user=name
(292779:000031) | From : <PC_sip_device> 31024@oxe-ov.alcatel.fr:5060 ; user=name
(292779:000032) | To : <"31004"> 31004@oxe-ov.alcatel.fr:5060 ; user=name
(292779:000033) +------------------------------------------------------------+
(292779:000034) | SDP :
(292779:000035) |  @IP:port = 135.118.226.39:7888
(292779:000036) |  ALGOS :
(292779:000037) |    PCMA
(292779:000038) |    G729
(292779:000039) |    DTMF : 101
(292779:000040) |  DIRECTION : SEND & RECEIVE
(292779:000041) | cac : false
(292779:000042) | Prack_Required: 0
(292779:000043) | Allow_UPDATE: 0
```

All the information received on the Call handling are given by the SIPMOTOR, the SIPMOTOR has already done an analyse and a treatment of these information.

We can see the “neqt” used to make the link between the SIPMOTOR trace and Call Handling trace (here 2250)

- The Call Handling checks the payload received.

```
(292779:000046) ctrl_payloads_on_reception_sdp payloads_recu[0]=0
(292779:000047) ctrl_payloads_on_reception_sdp payloads_recu[1]=17
(292779:000048) ctrl_payloads_on_reception_sdp dtmf_payload 101
```

- When a call is using a SIP Trunk Group, the call is treated throught this SIP Trunk Group like a call a on a T2 Trunk Group.

The Call Handling generates a SETUP message with the information given on the INVITE, the SETUP is different if the Trunk Group is ISDN or ABCF.

```
_____________________________________________________________________
| (292779:000128) Concatenated-Physical-Event :
| long: 177  desti: 0  source: 0  cryst: 19  cpl: 0  us: 0  term: 0  type a5
| tei: 0  <<<<  message sent : SETUP [05]    Call ref :  00 15
|               SENDING COMPLETE
|_____________________________________________________________________
|
| IE:[04] BEARER_CAPABILITY (l=3) 80 90 a3
| IE:[18] CHANNEL (l=2)  a0 90 -> T2 : No B channel
| IE:[1c] FACILITY (l=84)
|   [91] Discriminator of supplementary service applications
|   [aa] NFE (l=6):
|         [80] Source Entity (l=1) End_PTNX
|         [82] Destination Entity (l=1) End_PTNX
|   [8b] Interpretation APDU (l=1):  DISCARD (0)
|   [a1] INVOKE (l=25):
|        Invoke Ident. : 2ee0 (12000)
|         OP:  ECMA RO_CALLING_NAME (0)
|         [80] Name presentation allowed (l=13) 'PC_sip_device'
|   [a1] INVOKE (l=43):
|        Invoke Ident. : 0001 (1)
|         OP:  ALCATEL RO_CLASSMARKS (1)
|         [30] Sequence (l=30)
|            [80] Feature identifier (l=5)  06 04 70 1f 20
|            [82] Cug (l=1)  00
|            [ab] Sequence of Project data (l=18)
|               [30] Sequence (l=16)
|                  OP  :RO_CLASSMARKS_SUPPLEMENTARY_INFO_1 (134623475)
|                  [30] Sequence (l=10)
|                     [80] Trunk group feature (l=5) 06 00 00 20 04
|                     [83] Current entity (l=1) 01
```

When the SIP message is from the SIPMOTOR to the Call Handling, the direction is “message sent”.

On this setup all the information are present:

- The calling and called number
- The codecs
- The RTP connection information
- ...

The Call Ref is identical for outgoing and incoming messages (here Call ref : 00 15).

- The “CALL PROC” is present.

```
_____________________________________________________________________
| (292779:000291) Concatenated-Physical-Event :
| long: 22  desti: 0  source: 0  cryst: 19  cpl: 0  us: 0  term: 0  type a5
| tei: 0  >>>>  message received : CALL PROC (02)  Call ref :  00 15
|_____________________________________________________________________
|
| IE:[18] CHANNEL (l=2)  a0 90 -> T2 : No B channel
|_____________________________________________________________________
```

- The “ALERT” is generated for this call.

```
_____________________________________________________________________
| (292779:000294) Concatenated-Physical-Event :
| long: 101  desti: 0  source: 0  cryst: 19  cpl: 0  us: 0  term: 0  type a5
| tei: 0  >>>>  message received : ALERT (01)  Call ref :  00 15
|_____________________________________________________________________
|
| IE:[1c] FACILITY (l=64)
|   [91] Discriminator of supplementary service applications
|   [aa] NFE (l=6):
|         [80] Source Entity (l=1) End_PTNX
|         [82] Destination Entity (l=1) End_PTNX
|   [8b] Interpretation APDU (l=1):  DISCARD (0)
|   [a1] INVOKE (l=28):
|        Invoke Ident. : 2ee1 (12001)
|         OP:  ECMA RO_CALLED_NAME (1)
|         [80] Name presentation allowed (l=16) 'IPtouch 172.27.1'
|   [a1] INVOKE (l=20):
|        Invoke Ident. : 0001 (1)
|         OP:  ALCATEL RO_CLASSMARKS (1)
|         [30] Sequence (l=7)
|            [80] Feature identifier (l=5)  06 44 7e 1f 04
| IE:[1e] PROGRESS ID (l=2) 80 88
```

The ALERT has no RTP information, because the SDP on 18x is not set to true.

- The “ALERT” is transformed on a SIP message to the SIPMOTOR, but first the Call Handling select the good “neqt” to send the message to the SIPMOTOR.

```
(292779:000321) SIP : [send_to_motor] ipcSend resultat : 0 sur eqt : 2250
...
(292779:000323) +-------------------------------------------------------------+
(292779:000324) | Message sent UA (neqt : 2250-0) ----> SIP
(292779:000325) | Informational 180
(292779:000326) |RELATIVE REQUEST : INVITE
(292779:000327) | No SDP
(292779:000328) +-------------------------------------------------------------+
```

- The “CONNECT” is generated for this call.

```
_____________________________________________________________________
| (292789:000511) Concatenated-Physical-Event :
| long: 134  desti: 0  source: 0  cryst: 19  cpl: 0  us: 0  term: 0  type a5
| tei: 0  >>>>  message received : CONNECT (07)  Call ref :  00 15
|_____________________________________________________________________
|
| IE:[1c] FACILITY (l=64)
|   [91] Discriminator of supplementary service applications
|   [aa] NFE (l=6):
|         [80] Source Entity (l=1) End_PTNX
|         [82] Destination Entity (l=1) End_PTNX
|   [8b] Interpretation APDU (l=1):  DISCARD (0)
|   [a1] INVOKE (l=28):
|        Invoke Ident. : 2ee2 (12002)
|         OP:  ECMA RO_CONNECTED_NAME (2)
|         [80] Name presentation allowed (l=16) 'IPtouch 172.27.1'
|   [a1] INVOKE (l=20):
|        Invoke Ident. : 0001 (1)
|         OP:  ALCATEL RO_CLASSMARKS (1)
|         [30] Sequence (l=7)
|            [80] Feature identifier (l=5)  06 44 7e 1f 04
| IE:[4c] CONNECTED_NUMBER (l=7)  ->  00  81  Num : 31004
| [95] Locking shift. codeset : 5
| IE:[32] EI_PARTY_CATEGORY (l=1)  -> EXTENSION (1)
| [9f] Non-locking shift. codeset : 7
| IE:[06] EI_IP_PAYLOADS (l=1) -> G729 Ece 1 Vad 0
| [9f] Non-locking shift. codeset : 7
| IE:[0a] EI_RTP_INFO (l=30)
|    -> stop_packet=0 stop_rtp=0 h323=0 wc=0 rf=0 udp=1 rqm=0
|    -> Transm_Bande=1 detection_Q23=1 dtmf_payload=101
|    -> Port RTP     = 32514, IPv4 :   172.   27.  142.   64.
|    -> Port RTCP SR = 32515, IPv4 :   172.   27.  142.   64.
|    -> Port RTCP RR = 32515, IPv4 :   172.   27.  142.   64.
|    -> Port Fax     = 0, IPv4 :     0.    0.    0.    0.
```

The “CONNECT” has RTP information, these RTP information will be used to create the SDP.

- The “CONNECT” is transformed on a SIP message to the SIPMOTOR, but first the Call Handling select the good “neqt” to send the message to the SIPMOTOR.

```
(292789:000552) SIP : [send_to_motor] ipcSend resultat : 0 sur eqt : 2250
...
(292789:000554) +------------------------------------------------------------+
(292789:000555) | Message sent UA (neqt : 2250-0) ----> SIP
(292789:000556) | Successful 200
(292789:000557) |RELATIVE REQUEST : INVITE
(292789:000558) +------------------------------------------------------------+
(292789:000559) | SDP :
(292789:000560) |  @IP:port = 172.27.142.64:32514
(292789:000561) |  ALGOS :
(292789:000562) |    G729
(292789:000563) |    DTMF : 101
(292789:000564) |  DIRECTION : SEND & RECEIVE
(292789:000565) | AssertedAddress : <IPtouch 172.27.1> 31004@oxe-ov.alcatel.fr:5060
(292789:000566) | COLP
(292789:000567) +------------------------------------------------------------+
```

- The SIPMOTOR receives the ACK from the remote SIP equipment, and this message.

```
(292794:000580) +------------------------------------------------------------+
(292794:000581) | Message received SIP ----> UA (neqt : 2250)
(292794:000582) | ACK
(292794:000583) +------------------------------------------------------------+
```

- The ACK is transformed on a “CONNECT ACK”

```
 _____________________________________________________________________
| (292794:000586) Concatenated-Physical-Event :
| long: 18  desti: 0  source: 0  cryst: 19  cpl: 0  us: 0  term: 0  type a5
| tei: 0  <<<<  message sent : CONNECT ACK (0f)  Call ref :  00 15
|_____________________________________________________________________________
```

After call establishment, the call can be released by the OXE or by the remote SIP equipment.

- <u>Call released by the Call Handling:</u>
  - The “DISCONNECT” is generated on the call.

```
 _____________________________________________________________________
| (292810:000672) Concatenated-Physical-Event :
| long: 23  desti: 0  source: 0  cryst: 19  cpl: 0  us: 0  term: 0  type a5
| tei: 0  >>>>  message received : DISCONNECT [45]  Call ref :  00 15
|_____________________________________________________________________________
|
| IE:[08] CAUSE (l=3) 80 90 80 -> [90] NORMAL CALL CLEARING
|_____________________________________________________________________________
```

- The “DISCONNECT” is transformed on a SIP message to the SIPMOTOR, but first the Call Handling select the good “neqt” to send the message to the SIPMOTOR.

```
(292810:000682) SIP : [send_to_motor] ipcSend resultat : 0 sur eqt : 2250
...
(292810:000684) +-------------------------------------------------------------+
(292810:000685) | Message sent UA (neqt : 2250-0) ----> SIP
(292810:000686) | BYE
(292810:000687) +-------------------------------------------------------------+
```

- Answer of the BYE recieved by the SIPMOTOR transmits to the Call Handling.

```
(292811:000692) +-------------------------------------------------------------+
(292811:000693) | Message received SIP ----> UA (neqt : 2250)
(292811:000694) | Successful 200
(292811:000695) |RELATIVE REQUEST : BYE
(292811:000696) | No SDP
(292811:000697) +-------------------------------------------------------------+
```

- Answer of the BYE is transformed to a Call Handling message by a “RELEASE”.

```
_____________________________________________________________________________
| (292811:000699) Concatenated-Physical-Event :
| long: 23  desti: 0  source: 0  cryst: 19  cpl: 0  us: 0  term: 0  type a5
| tei: 0  <<<<  message sent : RELEASE [4d]   Call ref :  00 15
|_____________________________________________________________________________
|
| IE:[08] CAUSE (l=3) 80 90 80 -> [90] NORMAL CALL CLEARING
|_____________________________________________________________________________
```

Acknowledge of the “RELEASE” by a “REL COMP”.

```
_____________________________________________________________________________
| (292811:000705) Concatenated-Physical-Event :
| long: 23  desti: 0  source: 0  cryst: 19  cpl: 0  us: 0  term: 0  type a5
| tei: 0  >>>>  message received : REL COMP [5a]  Call ref :  00 15
|_____________________________________________________________________________
|
| IE:[08] CAUSE (l=3) 80 90 80 -> [90] NORMAL CALL CLEARING
|_____________________________________________________________________________
```

After the “REL COMP”, the call is completely ended on Call Handling side.

According to the problem, more options can be used on the Call Handling trace, due to them, more information are displayed. Here it is an example with the minimum of options to see the exchanges between the SIPMOTOR and the Call Handling.

It is important to understand the link between SIPMOTOR trace and Call Handling trace to make a minimum of analyses before to open a Service Request.

11.9.3 Incoming SIP call in case of SIP extension: SIPMOTOR point of view

Here an example of incoming call from a SIP extension to an IPtouch.

```
Tue Jun 26 08:03:05 2012 RECEIVE MESSAGE FROM NETWORK (135.118.226.21:61618 [UDP])
----------------------utf8-----------------------
INVITE sip:31004@oxe-ov.alcatel.fr SIP/2.0
Via: SIP/2.0/UDP 135.118.226.21:61618;branch=z9hG4bK-d87543-9c72747c0d38bb69-1--d87543-;rport
Max-Forwards: 70
Contact: <sip:31023@135.118.226.21:61618>
To: "31004"<sip:31004@oxe-ov.alcatel.fr>
From: "PC_sip_extenstion"<sip:31023@oxe-ov.alcatel.fr>;tag=c850be7c
Call-ID: MzBlMzgzNjY5NDg2NmE0NTRiMGYyYjMyOThjZmY4MWU.
CSeq: 1 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO
Content-Type: application/sdp
User-Agent: SIP Phone
Content-Length: 317

v=0
o=- 5 2 IN IP4 135.118.226.21
s=SIP Phone
c=IN IP4 135.118.226.21
t=0 0
m=audio 46194 RTP/AVP 8 18 101
a=fmtp:18 annexb=no
a=fmtp:101 0-15
a=rtpmap:101 telephone-event/8000
a=sendrecv
------------------------------------------------
```

The information “RECEIVE MESSAGE FROM NETWORK (135.118.226.21:61618[UDP])” is important to know that the call is an incoming one from the SIP equipment 135.118.226.21 in UDP.

- The OXE checks the Call-Id to know if this INVITE is an INVITE or a REINVITE.

```
Tue Jun 26 08:03:05 2012 11ef [CCall::getDialog] Confirmed Dialog is not found (ID = ;c850be7c)
Tue Jun 26 08:03:05 2012 11ef [CCall::getDialog] Initial Dialog Server not found
```

Here, it is an INVITE, because the dialog is not found.

- The transaction and the dialog are put in place.

```
Tue Jun 26 08:03:05 2012 210c [CTransaction::changeState] STATE CHANGED TO INITIAL
...
Tue Jun 26 08:03:05 2012 15fd [CDialog::onTransactionState(pTrans = 210c, previousState = Terminated, currentState
= Initial, reason = None]
```

Here, the transaction reference is “210c” and the dialog reference is “15fd”.

- The transaction status is changed, because the dialog is initiated.

```
Tue Jun 26 08:03:05 2012 210c [CTransaction::changeState] STATE CHANGED TO PROCEEDING
Tue Jun 26 08:03:05 2012 210c [CTransaction::changeState] notifying the parent dialog
```

When a transaction is linked to a dialog, the transaction changed from INITIAL to PROCEEDING.

- The SIPMOTOR generates the 100 Trying.

```
Tue Jun 26 08:03:05 2012 SEND MESSAGE TO NETWORK (135.118.226.21:61618 [UDP]) (BUFF LEN = 350)
----------------------utf8-----------------------
SIP/2.0 100 Trying
To: "31004" <sip:31004@oxe-ov.alcatel.fr>
From: "PC_sip_extenstion" <sip:31023@oxe-ov.alcatel.fr>;tag=c850be7c
Call-ID: MzBlMzgzNjY5NDg2NmE0NTRiMGYyYjMyOThjZmY4MWU.
CSeq: 1 INVITE
Via: SIP/2.0/UDP 135.118.226.21:61618;received=135.118.226.21;branch=z9hG4bK-d87543-9c72747c0d38bb69-1--d87543-
;rport=61618
Content-Length: 0
-------------------------------------------------
```

The 100 Trying is generated by the SIPMOTOR.

- The SIPMOTOR checks the Session Timer for the call.

```
Tue Jun 26 08:03:05 2012 [CSessionTimerContext::CSessionTimerContext] New CSessionTimerContext from request
(Server, UA)
Tue Jun 26 08:03:05 2012 [CSessionTimerContext::updateAfterRefreshReception] Update CSessionTimerContext (refresh
reception)
Tue Jun 26 08:03:05 2012 [CSessionTimerContext::updateSessionExpires] Session-Expires updated : 0
Tue Jun 26 08:03:05 2012 [CSessionTimerContext::setRefreshMethod] Allow refreshMethod=INVITE
```

In this case, the SIP equipment doesn't send "Session timer" information because the value is 0 (updated : 0).

- The SIPMOTOR makes the link between the transaction, the branch and the Cseq number.

```
Tue Jun 26 08:03:05 2012 15fd [CDialog::addTransaction] added transaction 210c with branch z9hG4bK-d87543-
9c72747c0d38bb69-1--d87543-, with CSeq 1
```

The "branch" is a parameter added to the "via" to identify it. Regarding rfc3261, all the branch values must start with "z9hG4bK".

The CSeq is used to indentify and to order a transaction, it consists of a sequence number and a method.

- The SIPMOTOR checks for which OXE equipment the call is from.

```
Tue Jun 26 08:03:05 2012 [isDomainFromGwExt] Host from request is : 172.27.141.151.
Tue Jun 26 08:03:05 2012 [isDomainFromGwExt] User from request is : 31023
Tue Jun 26 08:03:05 2012 [domain not from an External Gateway.
Tue Jun 26 08:03:05 2012 11ef[CMotorCall::setFilterUsedMode] To be traced = 0
Tue Jun 26 08:03:05 2012 11ef[CMotorCall::initOfUserType] values are reseted
Tue Jun 26 08:03:05 2012 [getFromHeader] displayName="PC_sip_extenstion".
Tue Jun 26 08:03:05 2012 [getFromHeader] =31023@oxe-ov.alcatel.fr.
Tue Jun 26 08:03:05 2012 [getFromHeader] clirPresent=0.
Tue Jun 26 08:03:05 2012 [isAddrInDico] user=31023 host=oxe-ov.alcatel.fr
Tue Jun 26 08:03:05 2012 [isUserInDico] 31023@oxe-ov.alcatel.fr
Tue Jun 26 08:03:05 2012 [isUserInDico]  found in the dictionnary.
Tue Jun 26 08:03:05 2012 [isAddrInDico] seplos station OK
```

Here, we can see that the call is from a SEPLOS station.

- The SIPMOTOR checks the number of licenses available.

```
Tue Jun 26 08:03:05 2012 11ef[CMotorCall::methodInviteReceived] nb available licenses=25
```

Here the number of licenses is 25, that means, 25 calls are possible on SIP using a SIP Trunk Group or SEPLOS users

- The SIPMOTOR checks if the IP address recieved is managed on an IP domain.

```
Tue Jun 26 08:03:05 2012    The recevied host 135.118.226.21
Tue Jun 26 08:03:05 2012  Trying to find the ip address in domain list
Tue Jun 26 08:03:05 2012  The entry dom : 142 add_type=1
Tue Jun 26 08:03:05 2012  The entry dom ip low :172.27.141.165
Tue Jun 26 08:03:05 2012  The entry ipaddress from low :135.118.226.21
Tue Jun 26 08:03:05 2012  The entry compare :1
Tue Jun 26 08:03:05 2012  The entry compare 2 :0
Tue Jun 26 08:03:05 2012  iplink_is_good_range_for_reg
...
Tue Jun 26 08:03:05 2012  The user domain is  142
```

Here, the IP address of the SIP equipment corresponds to the IP domain 142.

If the IP address doesn't match an IP domain, the SIPMOTOR returns:

```
Tue Jun 26 08:03:05 2012  The user is ipadd  not in any Domain range return state as -1
```

- The SIPMOTOR checks the SDP received on the INVITE.

```
Tue Jun 26 08:03:05 2012 [checkSdpValidity] Media 0 type 1 contains 3 formats.
Tue Jun 26 08:03:05 2012 [checkSdpValidity] Format : 8.
Tue Jun 26 08:03:05 2012 11ef[CMotorCall::isCryptoAuthorized] user crypto=0.
Tue Jun 26 08:03:05 2012 [convertSdpIntoTsdp] No Direction in the session part.
Tue Jun 26 08:03:05 2012 [convertSdpIntoTsdp] Check the direction in Session part - result:0.
Tue Jun 26 08:03:05 2012 [convertSdpIntoTsdp] media AUDIO detected (previous crypto=0).
Tue Jun 26 08:03:05 2012 [convertAudioMedia] The audio media contains  3 format(s).
Tue Jun 26 08:03:05 2012 [convertAudioMedia] Format 0 is 8.
```

The SDP contains in this SDP three formats of medias (8, 18 and 101), the direction is “sendrecv” meaning in both direction and the IP address of connection is 135.118.226.21.

- The message to Call Handling is prepared and sent to it.

```
Tue Jun 26 08:03:05 2012 11ef[CMotorCall::sendLgEvtSipCreate] Event sent on eqt : 2066
Tue Jun 26 08:03:05 2012 ** SEPLOS **
Tue Jun 26 08:03:05 2012 [display_ipc_out] ------------ Begin ---------------
Tue Jun 26 08:03:05 2012 Id : -1
Tue Jun 26 08:03:05 2012 INVITE
Tue Jun 26 08:03:05 2012 REQUEST URI : <> 31004@oxe-ov.alcatel.fr:5060 ; user=name
Tue Jun 26 08:03:05 2012 FROM : <PC_sip_extenstion> 31023@oxe-ov.alcatel.fr:5060 ; user=name
Tue Jun 26 08:03:05 2012 TO : <"31004"> 31004@oxe-ov.alcatel.fr:5060 ; user=name
Tue Jun 26 08:03:05 2012 CAC : 0
Tue Jun 26 08:03:05 2012   CAC ADDRESS :
Tue Jun 26 08:03:05 2012 CAC-CSBU info : UNKNOWN
Tue Jun 26 08:03:05 2012 CLIR : 0
Tue Jun 26 08:03:05 2012 Prack Required : 0
Tue Jun 26 08:03:05 2012 Allow Update : 0
Tue Jun 26 08:03:05 2012 SDP :
Tue Jun 26 08:03:05 2012   ADDRESS : 135.118.226.21 :46194
Tue Jun 26 08:03:05 2012   ALGOS :
Tue Jun 26 08:03:05 2012     PCMA
Tue Jun 26 08:03:05 2012     G729
Tue Jun 26 08:03:05 2012     101
Tue Jun 26 08:03:05 2012   DIRECTION : SEND & RECEIVE
Tue Jun 26 08:03:05 2012   crypto index : 0
Tue Jun 26 08:03:05 2012  N_GW_EXT : -1
Tue Jun 26 08:03:05 2012 [display_ipc_out] ------------- End ----------------
```

The call is sent to the Call handling on neqt 2066, regarding the type of SIP equipment detected by the SIPMOTOR, some information are added or not on this message.

- All the information about this call are sent to the Stand-By CPU.

```
Tue Jun 26 08:03:05 2012 SendToSipgwCpuSec: Message sent to the STAND-BY CPU
Tue Jun 26 08:03:05 2012 [receiveInviteMessage] send RemoteSdp to the StandBy.
Tue Jun 26 08:03:05 2012 SendToSipgwCpuSec: Message sent to the STAND-BY CPU
```

The information are sent to the Stand-By, like this, in case of bascul the SIP call will not be lost and known on the new main CPU

- The Call handling send back an answer for this INVITE.

```
Tue Jun 26 08:03:05 2012 [display_ipc_in] ------------ Begin ---------------
Tue Jun 26 08:03:05 2012 neqt : 2066 Id : 1
Tue Jun 26 08:03:05 2012 INFORMATIONAL
Tue Jun 26 08:03:05 2012 xx :        0
Tue Jun 26 08:03:05 2012   RELATIVE REQUEST : INVITE
Tue Jun 26 08:03:05 2012 [display_ipc_in] ------------- End ----------------
```

A “100 Trying” is sent by the Call Handling , but ignored by the SIPMOTOR.

```
Tue Jun 26 08:03:05 2012 [onIncomingEvent] INFORMATIONAL arrived.
Tue Jun 26 08:03:05 2012 [onIncomingEvent] 100 TRYING ignored.
```

This 100 Trying generated by the Call Handling is used by it to assign a “session” number for this call on the Call Handling side, but not used by the SIPMOTOR

- The Call handling sends back an answer for this INVITE.

```
Tue Jun 26 08:03:05 2012 [display_ipc_in] ------------ Begin ---------------
Tue Jun 26 08:03:05 2012 neqt : 2066 Id : 1
Tue Jun 26 08:03:05 2012 INFORMATIONAL
Tue Jun 26 08:03:05 2012 xx :       80
Tue Jun 26 08:03:05 2012  RELATIVE REQUEST : INVITE
Tue Jun 26 08:03:05 2012 SDP :
Tue Jun 26 08:03:05 2012   ADDRESS : 172.27.143.131 :32584
Tue Jun 26 08:03:05 2012   ALGOS :
Tue Jun 26 08:03:05 2012     G729
Tue Jun 26 08:03:05 2012     101
Tue Jun 26 08:03:05 2012   DIRECTION : SEND & RECEIVE
Tue Jun 26 08:03:05 2012   crypto index : 0
Tue Jun 26 08:03:05 2012 [display_ipc_in] ------------- End ----------------
```

A “180 Ringing” is sent by the Call Handling with SDP, for the moment, on a 18X message, the Call Handling will put everytime a SDP, no possibility to disable it.

- The SIPMOTOR checks if the SDP given is compatible with the SDP received in the INVITE.

```
1340690585 -> Tue Jun 26 08:03:05 2012 11ef[CMotorCall::makeResponseSdp] Audio media.
Tue Jun 26 08:03:05 2012 11ef[CMotorCall::appendAudioAttributToMedia] Direction: 0.
Tue Jun 26 08:03:05 2012 11ef[CMotorCall::appendAudioAttributToMedia] format 101
Tue Jun 26 08:03:05 2012 11ef[CMotorCall::makeResponseSdp] fromSdp.getMediaDesciprionCount :1
Tue Jun 26 08:03:05 2012 [sameCodec] accepted Format : 18.
Tue Jun 26 08:03:05 2012 [sameCodec] requested Format : 8.
Tue Jun 26 08:03:05 2012 [sameCodec] requested Format : 18.
Tue Jun 26 08:03:05 2012 [sameCodec] same Format.
Tue Jun 26 08:03:05 2012 11ef[CMotorCall::mediaAccepted] Media accepted: m=audio 32584 RTP/AVP 18 101
a=sendrecv
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
```

The codecs from the INVITE were 8 and 18, on the answer we have 18, in that case the call is accepted by SIPMOTOR for SDP point of view.

- The Call handling sends back an answer for this INVITE.

```
1340690585 -> Tue Jun 26 08:03:05 2012 SEND MESSAGE TO NETWORK (135.118.226.21:61618 [UDP]) (BUFF LEN = 827)
----------------------utf8-----------------------
SIP/2.0 180 Ringing
Allow: INVITE, ACK, CANCEL, BYE, PRACK, NOTIFY, REFER, SUBSCRIBE, OPTIONS, UPDATE
Contact: sip:oxe-ov.alcatel.fr
User-Agent: OmniPCX Enterprise R10.0 j1.410.45
Content-Type: application/sdp
To: "31004" <sip:31004@oxe-ov.alcatel.fr>;tag=05b5888d18d4e78f3554a55dadeefb08
From: "PC_sip_extenstion" <sip:31023@oxe-ov.alcatel.fr>;tag=c850be7c
Call-ID: MzBlMzgzNjY5NDg2NmE0NTRiMGYyYjMyOThjZmY4MWU.
CSeq: 1 INVITE
Via: SIP/2.0/UDP 135.118.226.21:61618;received=135.118.226.21;branch=z9hG4bK-d87543-9c72747c0d38bb69-1--d87543-
;rport=61618
Content-Length: 243

v=0
o=OXE 1340690585 1340690585 IN IP4 172.27.141.151
s=abs
c=IN IP4 172.27.143.131
t=0 0
m=audio 32584 RTP/AVP 18 101
a=sendrecv
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=ptime:20
a=maxptime:40
a=rtpmap:101 telephone-event/8000
------------------------------------------------
```

- For each SIP call event, a message is send to the Stand-By CPU.

```
Tue Jun 26 08:03:05 2012 [receiveInformationalEvent] UpdateContext send on the StandBy.
```

- The Call handling sends a new answer for this INVITE.

```
Tue Jun 26 08:03:08 2012 [display_ipc_in] ------------ Begin ---------------
Tue Jun 26 08:03:08 2012 neqt : 2066 Id : 1
Tue Jun 26 08:03:08 2012 SUCCESSFUL
Tue Jun 26 08:03:08 2012 xx :        0
Tue Jun 26 08:03:08 2012  RELATIVE REQUEST : INVITE
Tue Jun 26 08:03:08 2012 CLIR : 0
Tue Jun 26 08:03:08 2012 COLP : 1
Tue Jun 26 08:03:08 2012 CAC-CSBU info : UNKNOWN
Tue Jun 26 08:03:08 2012 SDP :
Tue Jun 26 08:03:08 2012   ADDRESS : 172.27.142.64 :32514
Tue Jun 26 08:03:08 2012   ALGOS :
Tue Jun 26 08:03:08 2012     G729
Tue Jun 26 08:03:08 2012     101
Tue Jun 26 08:03:08 2012   DIRECTION : SEND & RECEIVE
Tue Jun 26 08:03:08 2012   crypto index : 0
Tue Jun 26 08:03:08 2012 [display_ipc_in] ------------- End ----------------
```

A “200 ok” is sent to the SIPMOTOR with SDP

- The SIPMOTOR checks if the SDP given is compatible with the SDP received in the INVITE.

```
1340690588 -> Tue Jun 26 08:03:08 2012 11ef[CMotorCall::makeResponseSdp] Audio media.
Tue Jun 26 08:03:08 2012 11ef[CMotorCall::appendAudioAttributToMedia] Direction: 0.
Tue Jun 26 08:03:08 2012 11ef[CMotorCall::appendAudioAttributToMedia] format 101
Tue Jun 26 08:03:08 2012 11ef[CMotorCall::makeResponseSdp] fromSdp.getMediaDesciprionCount :1
Tue Jun 26 08:03:08 2012 [sameCodec] accepted Format : 18.
Tue Jun 26 08:03:08 2012 [sameCodec] requested Format : 8.
Tue Jun 26 08:03:08 2012 [sameCodec] requested Format : 18.
Tue Jun 26 08:03:08 2012 [sameCodec] same Format.
Tue Jun 26 08:03:08 2012 11ef[CMotorCall::mediaAccepted] Media accepted: m=audio 32514 RTP/AVP 18 101
a=sendrecv
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=ptime:20
a=maxptime:40
a=rtpmap:101 telephone-event/8000
.
```

The codecs from the INVITE were 8 and 18, on the answer we have 18, in that case the call is accepted by SIPMOTOR for SDP point of view.

- The SIPMOTOR is changing the status of the dialog.

```
Tue Jun 26 08:03:08 2012 15fd [CDialog::createResponse] create a CONFIRMED dialog
```

Due to this, the dialog reference and transaction reference are changed (internal SIPMOTOR functionning).

```
Tue Jun 26 08:03:08 2012 15fe [CDialog::CDialog] look for the transaction #0, transaction key = z9hG4bK-d87543-
9c72747c0d38bb69-1--d87543-
Tue Jun 26 08:03:08 2012 15fe [CDialog::CDialog] copy the transaction #0, transaction key = z9hG4bK-d87543-
9c72747c0d38bb69-1--d87543-
Tue Jun 26 08:03:08 2012 210d [CTransaction::CTransaction] Transaction is cloned in 4 state
```

The dialog reference is changed form “15fd” to “15fe”.
The transaction reference is changed from “210c” to “210d”.

- The SIPMOTOR is changing the stus of the dialog.

```
Tue Jun 26 08:03:08 2012 SEND MESSAGE TO NETWORK (135.118.226.21:61618 [UDP]) (BUFF LEN = 984)
----------------------utf8-----------------------
SIP/2.0 200 OK
Allow: INVITE, ACK, CANCEL, BYE, PRACK, NOTIFY, REFER, SUBSCRIBE, OPTIONS, UPDATE
Contact: sip:oxe-ov.alcatel.fr
Supported: replaces,timer,path,100rel
User-Agent: OmniPCX Enterprise R10.0 j1.410.45
Session-Expires: 1800;refresher=uas
P-Asserted-Identity: "IPtouch 172.27.142.64" <sip:31004@oxe-ov.alcatel.fr;user=phone>
Content-Type: application/sdp
To: "31004" <sip:31004@oxe-ov.alcatel.fr>;tag=05b5888d18d4e78f3554a55dadeefb08
From: "PC_sip_extenstion" <sip:31023@oxe-ov.alcatel.fr>;tag=c850be7c
Call-ID: MzBlMzgzNjY5NDg2NmE0NTRiMGYyYjMyOThjZmY4MWU.
CSeq: 1 INVITE
Via: SIP/2.0/UDP 135.118.226.21:61618;received=135.118.226.21;branch=z9hG4bK-d87543-9c72747c0d38bb69-1--d87543-
;rport=61618
Content-Length: 242

v=0
o=OXE 1340690585 1340690586 IN IP4 172.27.141.151
s=abs
c=IN IP4 172.27.142.64
t=0 0
m=audio 32514 RTP/AVP 18 101
a=sendrecv
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=ptime:20
a=maxptime:40
a=rtpmap:101 telephone-event/8000
------------------------------------------------
```

The 200ok sent to the remote SIP equipment is generated with information from the INVITE received and from the 200ok answer from the Call Handling.

- The SIPMOTOR is changing the status of the transaction.

```
Tue Jun 26 08:03:08 2012 210d [CTransProceedingState::createResponse] Final : Transaction changes to Completed
```

- The retransmission timers are started.

```
Tue Jun 26 08:03:08 2012 210d [CTransaction::startTimer] Timer G is started (delay = 500 ms)
Tue Jun 26 08:03:08 2012 210d [CTransaction::startTimer] Timer H is started (delay = 32000 ms)
```

- The SIPMOTOR receives a ACK for the 200ok.

```
Tue Jun 26 08:03:08 2012 RECEIVE MESSAGE FROM NETWORK (135.118.226.21:61618 [UDP])
----------------------utf8----------------------
ACK sip:oxe-ov.alcatel.fr SIP/2.0
Via: SIP/2.0/UDP 135.118.226.21:61618;branch=z9hG4bK-d87543-cc14ac1776189458-1--d87543-;rport
Max-Forwards: 70
Contact: <sip:31023@135.118.226.21:61618>
To: "31004"<sip:31004@oxe-ov.alcatel.fr>;tag=05b5888d18d4e78f3554a55dadeefb08
From: "PC_sip_extenstion"<sip:31023@oxe-ov.alcatel.fr>;tag=c850be7c
Call-ID: MzBlMzgzNjY5NDg2NmE0NTRiMGYyYjMyOThjZmY4MWU.
CSeq: 1 ACK
User-Agent: SIP Phone
Content-Length: 0
------------------------------------------------
```

- The SIPMOTOR is changing the status of the transaction.

```
Tue Jun 26 08:03:08 2012 210d [CTransaction::changeState] STATE CHANGED TO TERMINATED
```

- The retransmission timers are freed.

```
Tue Jun 26 08:03:08 2012 210d [CTransaction::freeTimerToken] Timer G is freed
Tue Jun 26 08:03:08 2012 210d [CTransaction::freeTimerToken] Timer H is freed
```

- The SIPMOTOR is changing the status of the dialog.

```
Tue Jun 26 08:03:08 2012 15fe [CDialog::receiveAckRequest] the INVITE request is terminated
```

- The ACK is sent to the Call Handling.

```
Tue Jun 26 08:03:08 2012 [display_ipc_out] ------------ Begin ---------------
Tue Jun 26 08:03:08 2012 Id : 1
Tue Jun 26 08:03:08 2012 ACK
Tue Jun 26 08:03:08 2012 [display_ipc_out] ------------- End ----------------
```

After call establishment, the call can be released by the OXE or by the remote SIP equipment.

- Call released by the OXE:

  - The BYE is sent from the Call Handling.

```
Tue Jun 26 08:03:10 2012 [display_ipc_in] ------------ Begin ---------------
Tue Jun 26 08:03:10 2012 neqt : 2066 Id : 1
Tue Jun 26 08:03:10 2012 BYE
Tue Jun 26 08:03:10 2012 [display_ipc_in] ------------- End ----------------
```

- Creation of a new transaction for the BYE.

```
Tue Jun 26 08:03:10 2012 2110 [CTransaction::changeState] STATE CHANGED TO INITIAL
```

The BYE is a new transaction for a SIP call, in that case, the transaction reference it is “2110”, and the status is “INITIAL”.

- The BYE is sent to the remote SIP equipment.

```
Tue Jun 26 08:03:10 2012 SEND MESSAGE TO NETWORK (135.118.226.21:61618 [UDP]) (BUFF LEN = 454)
----------------------utf8-----------------------
BYE sip:31023@135.118.226.21:61618 SIP/2.0
Supported: replaces,timer,path,100rel
User-Agent: OmniPCX Enterprise R10.0 j1.410.45
To: sip:31023@oxe-ov.alcatel.fr;tag=c850be7c
From: "31004" <sip:31004@oxe-ov.alcatel.fr>;tag=05b5888d18d4e78f3554a55dadeefb08
Call-ID: MzBlMzgzNjY5NDg2NmE0NTRiMGYyYjMyOThjZmY4MWU.
CSeq: 716266225 BYE
Via: SIP/2.0/UDP 172.27.141.151;branch=z9hG4bK2385fb34fcefc38c24fa6848df37e986
Max-Forwards: 70
Content-Length: 0
-------------------------------------------------
```

- The SIPMOTOR changes the transaction state.

```
Tue Jun 26 08:03:10 2012 2110 [CTransaction::changeState] STATE CHANGED TO TRYING
```

- The retransmission timers are started.

```
Tue Jun 26 08:03:10 2012 2110 [CTransaction::startTimer] Timer E is started (delay = 500 ms)
Tue Jun 26 08:03:10 2012 2110 [CTransaction::startTimer] Timer F is started (delay = 16000 ms)
```

- The 200ok of the BYE request is received from the remote SIP equipment.

```
Tue Jun 26 08:03:10 2012 RECEIVE MESSAGE FROM NETWORK (135.118.226.21:61618 [UDP])
----------------------utf8-----------------------
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.27.141.151;branch=z9hG4bK2385fb34fcefc38c24fa6848df37e986
Contact: <sip:31023@135.118.226.21:61618>
To: <sip:31023@oxe-ov.alcatel.fr>;tag=c850be7c
From: "31004"<sip:31004@oxe-ov.alcatel.fr>;tag=05b5888d18d4e78f3554a55dadeefb08
Call-ID: MzBlMzgzNjY5NDg2NmE0NTRiMGYyYjMyOThjZmY4MWU.
CSeq: 716266225 BYE
User-Agent: SIP Phone
Content-Length: 0
-------------------------------------------------
```

- The SIPMOTOR changes this transaction state.

```
Tue Jun 26 08:03:10 2012 2110 [CTransaction::changeState] STATE CHANGED TO COMPLETED
```

- The retransmission timers are freed.

```
Tue Jun 26 08:03:10 2012 2110 [CTransaction::freeTimerToken] Timer E is freed
Tue Jun 26 08:03:10 2012 2110 [CTransaction::freeTimerToken] Timer F is freed
```

- The 200ok of the BYE request is sent to the Call Handling.

```
Tue Jun 26 08:03:10 2012 ** SEPLOS **
Tue Jun 26 08:03:10 2012 [sendLgEvtSip] Event sent on eqt : 2066 Id :1
Tue Jun 26 08:03:10 2012 [display_ipc_out] ------------ Begin ---------------
Tue Jun 26 08:03:10 2012 Id : 1
Tue Jun 26 08:03:10 2012 SUCCESSFUL
Tue Jun 26 08:03:10 2012 xx :        0
Tue Jun 26 08:03:10 2012  RELATIVE REQUEST : BYE
Tue Jun 26 08:03:10 2012 CAC-CSBU info : UNKNOWN
```

- The Call Handling sent a message to the SIPMOTOR to release the “neqt” associated to this SIP call

```
Tue Jun 26 08:03:10 2012 [display_ipc_in] ------------ Begin ---------------
Tue Jun 26 08:03:10 2012 neqt : 2066 Id : 1
Tue Jun 26 08:03:10 2012 SIP EQT RELEASED
Tue Jun 26 08:03:10 2012 [display_ipc_in] ------------- End ----------------
```

- The SIPMOTOR acknowledge the release of the “neqt”

```
Tue Jun 26 08:03:10 2012 [CMotorCallManager::onIncomingEvent] The call with eqt: 2066 has released its equipment.
Tue Jun 26 08:03:10 2012 [CMotorCallManager::onIncomingEvent] state = TERMINATED_STATE.
Tue Jun 26 08:03:10 2012 11ef[CMotorCall::unRegister] Remove eqt : 2066 diag : 1 from the map.
Tue Jun 26 08:03:10 2012 [CMotorCallManager::eraseCallwithEqt] erase 2066 1.
```

- The SIPMOTOR kills the SIP call

```
Tue Jun 26 08:03:10 2012 [CMotorCallManager::onIncomingEvent] killSession.
Tue Jun 26 08:03:10 2012 11ef [CCall::killSession]
```

- The SIPMOTOR changes the state of the transactions

```
Tue Jun 26 08:03:10 2012 210c [CTransaction::changeState] STATE CHANGED TO TERMINATED
...
Tue Jun 26 08:03:10 2012 2110 [CTransaction::changeState] STATE CHANGED TO TERMINATED
```

- <u>Call released by the remote SIP equipment:</u>

- The BYE is received from the remote SIP equipment.

```
Tue Jun 26 08:03:10 2012 RECEIVE MESSAGE FROM NETWORK (135.118.226.21:61618 [UDP])
----------------------utf8-----------------------
BYE sip:oxe-ov.alcatel.fr SIP/2.0
Via: SIP/2.0/UDP 135.118.226.21:61618;branch=z9hG4bK-d87543-c47926131a084707-1--d87543-;rport
Max-Forwards: 70
Contact: <sip:31023@135.118.226.21:61618>
To: "31004"<sip:31004@oxe-ov.alcatel.fr>;tag=efa4b05316a486724541975cb22707d1
From: "PC_sip_extenstion"<sip:31023@oxe-ov.alcatel.fr>;tag=c55fb830
Call-ID: MzIwMmRjNGI3YTE3ZjkwZTE0ODE4Y2IzZGU1ZTdjZDM.
CSeq: 2 BYE
User-Agent: SIP Phone
Content-Length: 0
------------------------------------------------
```

- The SIPMOTOR checks if the dialog is already exist.

```
Tue Jun 26 08:03:10 2012 11ef [CCall::getDialog] Confirmed Dialog found
```

- Creation of a new transaction for the BYE.

```
Tue Jun 26 08:03:10 2012 2110 [CTransaction::changeState] STATE CHANGED TO INITIAL
```

The BYE is a new transaction for a SIP call, in that case, the transaction reference it is “21a7”, and the status is “INITIAL”.

- The SIPMOTOR changes the transaction state.

```
Tue Jun 26 08:03:10 2012 2110 [CTransaction::changeState] STATE CHANGED TO TRYING
```

- The BYE is sent to the Call handling.

```
Tue Jun 26 08:03:10 2012 [display_ipc_out] ------------ Begin ---------------
Tue Jun 26 08:03:10 2012 Id : -1
Tue Jun 26 08:03:10 2012 BYE
Tue Jun 26 08:03:10 2012 [display_ipc_out] ------------- End ----------------
```

- The Call Handling answers to the SIPMOTOR.

```
Tue Jun 26 08:03:10 2012 [display_ipc_in] ------------ Begin ---------------
Tue Jun 26 08:03:10 2012 neqt : 2266 Id : -1
Tue Jun 26 08:03:10 2012 SUCCESSFUL
Tue Jun 26 08:03:10 2012 xx :        0
Tue Jun 26 08:03:10 2012  RELATIVE REQUEST : BYE
Tue Jun 26 08:03:10 2012 CLIR : 0
Tue Jun 26 08:03:10 2012 COLP : 0
Tue Jun 26 08:03:10 2012 CAC-CSBU info : UNKNOWN
Tue Jun 26 08:03:10 2012 [display_ipc_in] ------------- End ---------------
```

- The SIPMOTOR sends the 200 ok of the BYE to the remote SIP equipment.

```
Tue Jun 26 08:03:10 2012 SEND MESSAGE TO NETWORK (135.118.226.39:25648 [UDP]) (BUFF LEN = 546)
----------------------utf8-----------------------
SIP/2.0 200 OK
Allow: INVITE, ACK, CANCEL, BYE, PRACK, NOTIFY, REFER, SUBSCRIBE, OPTIONS, UPDATE
Supported: replaces,timer,path,100rel
User-Agent: OmniPCX Enterprise R10.0 j1.410.45
To: "31004"<sip:31004@oxe-ov.alcatel.fr>;tag=efa4b05316a486724541975cb22707d1
From: "PC_sip_extenstion"<sip:31023@oxe-ov.alcatel.fr>;tag=c55fb830
Call-ID: MzIwMmRjNGI3YTE3ZjkwZTE0ODE4Y2IzZGU1ZTdjZDM.
CSeq: 2 BYE
Via: SIP/2.0/UDP 135.118.226.39:25648;received=135.118.226.39;branch=z9hG4bK-d87543-cf501c2f3311d050-1--d87543-
;rport=25648
Content-Length: 0
-------------------------------------------------
```

- The SIPMOTOR changes the transaction state.

```
Tue Jun 26 08:03:10 2012 2110 [CTransaction::changeState] STATE CHANGED TO COMPLETED
```

- The Call Handling sends a message to the SIPMOTOR to release the “neqt” associated to this SIP call

```
Tue Jun 26 08:03:10 2012 [display_ipc_in] ------------ Begin ---------------
Tue Jun 26 08:03:10 2012 neqt : 2266 Id : -1
Tue Jun 26 08:03:10 2012 SIP EQT RELEASED
Tue Jun 26 08:03:10 2012 [display_ipc_in] ------------- End ----------------
```

- The SIPMOTOR acknowledge the release of the “neqt”

```
Tue Jun 26 08:03:10 2012 [CMotorCallManager::onIncomingEvent] The call with eqt: 2066 has released its equipment.
Tue Jun 26 08:03:48 2012 [CMotorCallManager::onIncomingEvent] state = TERMINATED_STATE.
Tue Jun 26 08:03:48 2012 11fc[CMotorCall::unRegister] Remove eqt : 2066 diag : 1 from the map.
Tue Jun 26 08:03:48 2012 [CMotorCallManager::eraseCallwithEqt] erase 2066 1
```

- The SIPMOTOR kills the SIP call

```
Tue Jun 26 08:03:10 2012 [CMotorCallManager::onIncomingEvent] killSession.
Tue Jun 26 08:03:10 2012 11ef [CCall::killSession]
```

- The SIPMOTOR change the state of the transactions

```
Tue Jun 26 08:03:10 2012 210c [CTransaction::changeState] STATE CHANGED TO TERMINATED
...
Tue Jun 26 08:03:10 2012 2110 [CTransaction::changeState] STATE CHANGED TO TERMINATED
```

11.9.4 Incoming SIP call in case of SIP extension: Call Handling point of view

Here an example of incoming call from a SIP extension to an IPtouch.

Traces option used :

```
>tuner km
>tuner clear-traces
>trc i
>actdbg all=off
>tuner +cpu +cpl +at hybrid=on
>actdbg sip=on csip=on
>mtracer -a
```

- The call arrives on the SIPMOTOR, and sending to the Call Handling

```
(600095:000062) CSIP @@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@ 02066 activated @@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@
(600095:000063) CSIP_receiveSipMsg
(600095:000064) +------------------------------------------------------------+
(600095:000065) | Message received SIP ----> UA (neqt : 2066)
(600095:000066) | INVITE : 31004@oxe-ov.alcatel.fr:5060 ; user=name
(600095:000067) | From : <PC_sip_extenstion> 31023@oxe-ov.alcatel.fr:5060 ; user=name
(600095:000068) | To : <"31004"> 31004@oxe-ov.alcatel.fr:5060 ; user=name
(600096:000069) +------------------------------------------------------------+
(600096:000070) | SDP :
(600096:000071) |  @IP:port = 135.118.226.21:46194
(600096:000072) |  ALGOS :
(600096:000073) |    PCMA
(600096:000074) |    G729
(600096:000075) |    DTMF : 101
(600096:000076) |  DIRECTION : SEND & RECEIVE
(600096:000077) | cac : false
(600096:000078) | Prack_Required: 0
(600096:000079) | Allow UPDATE: 0
```

In case of SIP Extension, all The call Handling treatment for the call starts by the message “CSIP”, for SIP extension point of view.

On the first line, the information “02066 activated” is used to inform that the Call Handling starts the treatment of the SIP extension with the neqt 2066.

- The Call Handling checks if a session is already opened for this SIP extension user.

```
(600096:000087) ..CSIPMsgSipInvite::getSession
(600096:000088) ....CSIP_getSessionFromRequestURI
(600096:000089) ......Didn't retrieve session for requestUri 31004
(600096:000090) ....CSIP_getFreeSession
(600096:000091) ......Got free session 1 for ChId 80 CSIP_INVITE_WAIT_STATUS_CH_ID
```

In that case, no session opened, the Call Handling assigns to this call the session number 1, for a second call (if the first call is still up) the session will be 2, etc...

- The Call Handling generates a 100 Trying for this session

```
(600096:000094) ......CSIPSession#1ChId#80::sendSipInformational
(600096:000095) ........CSIPSession#1ChId#80::emitMsgToSIPMotor
(600096:000096) ..........SIP_INFORMATIONAL sent
(600096:000097) +------------------------------------------------------------+
(600096:000098) | Message sent UA (neqt : 2066-1) ----> SIP
(600096:000099) | Informational 100
(600096:000100) |RELATIVE REQUEST : INVITE
(600096:000101) | No SDP
(600096:000102) +------------------------------------------------------------+
```

This 100 Trying will not take into account by the SIPMOTOR, it is used only to start the session on the Call handling side.

- Getting the SDP information received

```
(600096:000121) CSIP_tradKey chId=128 CSIP_START_CALL
(600096:000122) CSIP_analyzeSdp 135.118.226.21:46194 DTMF=101 SIP_SENDRECV
(600096:000123)                 G_711_A/G_729_A -> G_711_A/G_729_A
(600096:000124) CSIP_tradKey -> cnx_create_tab(0, -1, 135.118.226.21:46194)
(600096:000125) CSIP_tradKey kindofkey=VSYST (6) cokey=17
(600096:000126) CSIP_sendInfoCs : No call server informations authorization.
```

This 100 Trying will not take into account by the SIPMOTOR, it is used only to start the session on the Call handling side.

- Analyse of the SDP information

```
 (600096:000136) put_rtp_info end 2066 local.wc=0 distant.wc=0
(600096:000137) sip_ems_with_rfc2833-->disa_for_remote_ext=0
(600096:000138) sip_ems_with_rfc2833-->Result=0
(600096:000139) Exist_RCL_link-->Result=0,dtmf_direction=1
(600096:000141) SIP: mise a jour VPN
(600096:000142) dtmf_to_vpn_from_abc : dtmf_payload(2066)=101
(600096:000143) dtmf_to_vpn_from_abc : !LIEN_VPN
(600096:000144) Marhaban bikom dans le monde SIP : dtmf_payload(2066)= 101
(600096:000145) CSIP_isNwkCallWithSeplos neqt 2066 abc -1 vpn -1 result 0
(600096:000146) is_ems_ext_gw-->neqt=2066,Result=0
(600096:000147) send_cpl_connect_rtp_direct-->dtmf_direction=1
(600096:000152) CSIP_sendUpdateMsgFromCh call_id=0->1 neqt=-1->2066 state=NO_SCREEN->SCREEN_DIAL_0_DIGIT
(600096:000153) CSIP_sendUpdateMsgFromCh -> cnx_create_tab(1, 2066)
(600096:000154) CSIP_constructDistantField UTF-8 SCREEN_DIAL_0_DIGIT key=1
(600096:000155)   ""
(600096:000156) CSIP_constructOtherField UTF-8 SCREEN_DIAL_0_DIGIT key=1
(600096:000157)   "PC" 31023
(600096:000158) CSIP_constructSdp Default case
(600096:000159)   172.27.143.131:32584 G_729_A DTMF=101 SIP_SENDRECV
(600096:000160) CSIP_constructOtherInfo clir=0 forward=0 autoAnswer=0
(600096:000161) ..CSIPMsgInFactory::makeMsgInCh
(600096:000162) ..new CSIPMsgChDial0Digit at 0x54038ce8 - counter 1
(600096:000163) CSIP_sendUpdateMsgFromCh -> call CSIP_setFeatureList
(600096:000164) nulog_final: 0 typconv : 0 ptdemi->forwarded_neqph:-1
(600096:000165) CSIP_setFeatureList
(600096:000168) CSIP_sendInfoCs : No call server informations authorization..
```

- The Call handling gets the SDP infomation of the equipment for the RBT to generate the SDP of the 180

```
(600096:000195) CSIP_sendInfoCs : No call server informations authorization.
(600096:000198) chgt_local_rtp_info ptdemi->info.hinfo=0 ptdemi->neqt=2066
(600096:000199) chgt_local_rtp_info local.wc=0 distant.wc=0 before update
(600096:000200) chgt_local_rtp_info end local.wc=0 distant.wc=0
(600096:000201) CSIP_sendInfoCs : No call server informations authorization.
(600096:000203) CSIP_sendUpdateMsgFromCh call_id=1->1 neqt=2066->2066 state=SCREEN_DIAL_0_DIGIT->SCREEN_DIAL_DIGIT
(600096:000204) CSIP_constructDistantField UTF-8 SCREEN_DIAL_DIGIT key=1
(600096:000205)   ""
(600096:000206) CSIP_constructOtherField UTF-8 SCREEN_DIAL_DIGIT key=1
(600096:000207)   " PC" 31023
(600096:000208) CSIP_constructSdp Default case
(600096:000209)   172.27.143.131:32584 G_729_A DTMF=101 SIP_SENDRECV
(600096:000210) CSIP_constructOtherInfo clir=0 forward=0 autoAnswer=0
(600096:000211) ..CSIPMsgInFactory::makeMsgInCh
(600096:000212) ..new CSIPMsgChDialDigit at 0x54038ce8 - counter 1
(600096:000213) CSIP_sendUpdateMsgFromCh -> call CSIP_setFeatureList
(600096:000214) nulog_final: 0 typconv : 0 ptdemi->forwarded_neqph:-1
(600096:000215) CSIP_setFeatureList
(600096:000216) CSIP_sendInfoCs : No call server informations authorization.
```

Here, the IP address for the RBT is 172.27.143.131, and the port used is 32584 and the codec used is G729 (this information appears few times on the trace)

- The 180 is generated by the Call Handling and sent to the SIPMOTOR.

```
(600096:000400) CSIP_receiveComAction
(600096:000401) ..activeChId 1 featureList --
(600096:000402) ..CSIP Queue CSIPMsgChCalledStatus
(600096:000403) ..CSIPMsgChCalledStatus::getSession
(600096:000404) ....CSIP_getSessionFromChId
(600096:000405) ......Retrieved session 1 for ChId 1
(600096:000406) ..CSIPMsgChCalledStatus::execute
```

- The “CALL PROC” is present.

The state of the session, for Call Handling point of view, change to “CSIPStateInvite180WaitConversation”

- The Call handling gets the SDP infomation of the equipment for the 200ok

```
neqttouc neqt=2049 nekip=2049 toucacod=1
(600121:000476) neqttouc result=1000801 en Hexa !!!
(600121:000477) neqttouc neqt=2049 nekip=2066 toucacod=1
(600121:000478) neqttouc result=1000812 en Hexa !!!
(600121:000479) sip_behind_ice-->neqt=2049,Result=0
(600121:000480) sip_behind_ice-->neqt=2066,Result=0
(600121:000486) SIP ipphone : interro statut 0 ptdemi->neqt(2049)
(600121:000487) SIP ipphone : GetneqtEnFace = -1 payload = 101 neqt =(2066)
(600121:000490) put_rtp_info end 2066 local.wc=0 distant.wc=0
(600121:000497) neqttouc neqt=2066 nekip=2049 toucacod=1
(600121:000498) neqttouc result=1000801 en Hexa !!!
(600121:000499) sip_behind_ice-->neqt=2066,Result=0
(600121:000500) sip_behind_ice-->neqt=2049,Result=0
(600121:000503) numunpack_trace: 31004
(600121:000504) from_same_nb_in_mes : nulog=27,numero_lg=5
(600121:000505) CSIP_msg_notify_management : No MWI subscription.
(600121:000506) sip_behind_ice-->neqt=2066,Result=0
(600121:000507) sip_behind_ice-->neqt=2049,Result=0
(600121:000510) CSIP_sendUpdateMsgFromCh call_id=1->1 neqt=2049->2049 state=SCREEN_CALLED_STATUS-
>SCREEN_CONVERSATIO
(600121:000511) CSIP_constructDistantField UTF-8 SCREEN_CONVERSATION key=1
(600121:000512)   "IPtouch 172.27.142.64" 31004
(600121:000513) CSIP_constructOtherField UTF-8 SCREEN_CONVERSATION key=1
(600121:000514)   "PC" 31023
(600121:000515) CSIP_constructSdp Default case
(600121:000516)   172.27.142.64:32514 G_729_A DTMF=101 SIP_SENDRECV
(600121:000517) CSIP_constructOtherInfo clir=0 forward=0 autoAnswer=0
(600121:000518) ..CSIPMsgInFactory::makeMsgInCh
(600121:000519) ..new CSIPMsgChConversation at 0x54038ce8 - counter 1
(600121:000520) CSIP_sendUpdateMsgFromCh -> call CSIP_setFeatureList
(600121:000521) nulog_final: 4 typconv : 1 ptdemi->forwarded_neqph:-1
(600121:000522) CSIP_setFeatureList START_CALL HOLD
(600121:000523) CSIP_sendInfoCs : No call server informations authorization.
```

Here, the IP address for the 200ok is 172.27.142.64, and the port used is 32514 and the codec used is G729, this SDP correponds to the IPtouch.

- The 200ok is generated by the Call Handling and sent to the SIPMOTOR

```
(600121:000525) CSIP_receiveComAction
(600121:000526) ..activeChId 1 featureList START_CALL HOLD
(600121:000527) ..CSIP Queue CSIPMsgChConversation
(600121:000528) ..CSIPMsgChConversation::getSession
(600121:000529) ....CSIP_getSessionFromChId
(600121:000530) ......Retrieved session 1 for ChId 1
(600121:000531) ..CSIPMsgChConversation::execute
(600121:000532) ....CSIPStateInvite180WaitConversation::doCSIPMsgChConversation
(600121:000533) ......CSIPSession#1ChId#1::setDistantSdp
(600121:000534) ........CSIPSession#1ChId#1::compareDistantSdp
(600121:000535) ..........Change 172.27.143.131:32584 G_729_A DTMF=101 SIP_SENDRECV
(600121:000536) ..........    -> 172.27.142.64:32514 G_729_A DTMF=101 SIP_SENDRECV
(600121:000537) ........CSIPSession#1ChId#1::resetIsSdpSentInInf
(600121:000538) ......CSIPSession#1ChId#1::setDistantClir
(600121:000539) ......CSIPSession#1ChId#1::setDistantName
(600121:000540) ......CSIPSession#1ChId#1::setDistantNumber
(600121:000541) ......CSIPSession#1ChId#1::sendSipSuccessful
(600121:000542) ........CSIPSession#1ChId#1::emitMsgToSIPMotor
(600121:000543) ..........SIP_SUCCESSFUL sent
(600121:000544) +------------------------------------------------------------+
(600121:000545) | Message sent UA (neqt : 2066-1) ----> SIP
(600121:000546) | Successful 200
(600121:000547) |RELATIVE REQUEST : INVITE
(600121:000548) +------------------------------------------------------------+
(600121:000549) | SDP :
(600121:000550) |  @IP:port = 172.27.142.64:32514
(600121:000551) |  ALGOS :
(600121:000552) |    G729
(600121:000553) |    DTMF : 101
(600121:000554) |  DIRECTION : SEND & RECEIVE
(600121:000555) | AssertedAddress : <IPtouch 172.27.142.64> 31004@oxe-ov.alcatel.fr:5060
(600121:000556) | COLP
(600121:000557) +------------------------------------------------------------+
(600121:000558) ......CSIPSession#1ChId#1::changeState
```

The state of the session, for Call Handling point of view, change to “CSIPStateConnectedWaitAck”.

- The ACK is received from the SIPMOTOR

```
 (600126:000641) CSIP_receiveSipMsg
(600126:000642) +------------------------------------------------------------+
(600126:000643) | Message received SIP ----> UA (neqt : 2066-1)
(600126:000644) | ACK
(600126:000645) +------------------------------------------------------------+
(600126:000646) ..activeChId 1 featureList START_CALL HOLD
(600126:000647) ..CSIPMsgInFactory::makeMsgInSip
(600126:000648) ....SIP_ACK dialogId 1
(600126:000649) ....new CSIPMsgSipAck at 0x54038f90 - counter 2
(600126:000650) ..CSIP Queue CSIPMsgSipAck < CSIPMsgChUpdateRtp
(600126:000651) ..CSIPMsgSipAck::getSession
(600126:000652) ....CSIP_getSessionFromId
(600126:000653) ......Retrieved session 1 with ChId 1
(600126:000654) ..CSIPMsgSipAck::execute
(600126:000655) ....CSIPStateConnectedWaitAck::doCSIPMsgSipAck
(600126:000656) ......CSIPSession#1ChId#1::changeState
(600126:000657) ........CSIPStateConnectedWaitAck -> CSIPStateConnected
```

The state of the session, for Call Handling point of view, change to “CSIPStateConnected”.

- <u>Call released by the OXE:</u>

  - The BYE is generated by the Call Handling and sent to the SIPMOTOR

```
(600143:000733) CSIP_receiveComAction
(600143:000734) ..activeChId 1 featureList HOLD
(600143:000735) ..CSIP Queue CSIPMsgChOnHook
(600143:000736) ..CSIPMsgChOnHook::getSession
(600143:000737) ....CSIP_getSessionFromChId
(600143:000738) ......Retrieved session 1 for ChId 1
(600143:000739) ..CSIPMsgChOnHook::execute
(600143:000740) ....CSIPStateConnected::doCSIPMsgChOnHook
(600143:000741) ......CSIPSession#1ChId#1::sendMsgToCh
(600143:000742) ........CSIP_HANG_UP
(600143:000743) ......CSIPSession#1ChId#1::sendSipBye
(600143:000744) ........CSIPSession#1ChId#1::emitMsgToSIPMotor
(600143:000745) ..........SIP_BYE sent
(600143:000746) +------------------------------------------------------------+
(600143:000747) | Message sent UA (neqt : 2066-1) ----> SIP
(600143:000748) | BYE
(600143:000749) +------------------------------------------------------------+
(600143:000750) ......CSIPSession#1ChId#1::changeState
(600143:000751) ........CSIPStateConnected -> CSIPStateByeWait200
```

The state of the session, for Call Handling point of view, change to “CSIPStateByeWait200”.

  - The 200OK of the BYE is received on the Call Handling

```
(600144:000831) CSIP_receiveSipMsg
(600144:000832) +------------------------------------------------------------+
(600144:000833) | Message received SIP ----> UA (neqt : 2066-1)
(600144:000834) | Successful 200
(600144:000835) |RELATIVE REQUEST : BYE
(600144:000836) | No SDP
(600144:000837) +------------------------------------------------------------+
(600144:000838) ..activeChId 0 featureList START_CALL
```

The state of the session, for Call Handling point of view, change to “CSIPStateIdle”.

- The “neqt” is released (SIP_EQT_RELEASED sent)
- The “half-com” is released (CSIP lib__demi() called for neqt 2066)

On the Call Handling, the SIP extension calls have a “session”, this is the evolution of the session state from the INVITE to the 200ok of the BYE:

- CSIPStateIdle -> CSIPStateInviteWaitDial0Digit
  - Changing state from the INVITE to the 100 Trying

- CSIPStateInviteWaitDial0Digit -> CSIPStateInviteWaitCalledStatus
  - Changing state from the 100 Trying to the 180 Ringing

- CSIPStateInviteWaitCalledStatus -> CSIPStateInvite180WaitConversation
  - Changing state from the 180 Ringing to the 200 Ok

- CSIPStateInvite180WaitConversation -> CSIPStateConnectedWaitAck
  - Changing state from the 200 Ok to the ACK

- CSIPStateConnectedWaitAck -> CSIPStateConnected
  - Changing state from the ACK to the BYE

- CSIPStateConnected -> CSIPStateByeWait200
  - Changing state from the BYE to the 200 Ok of the BYE

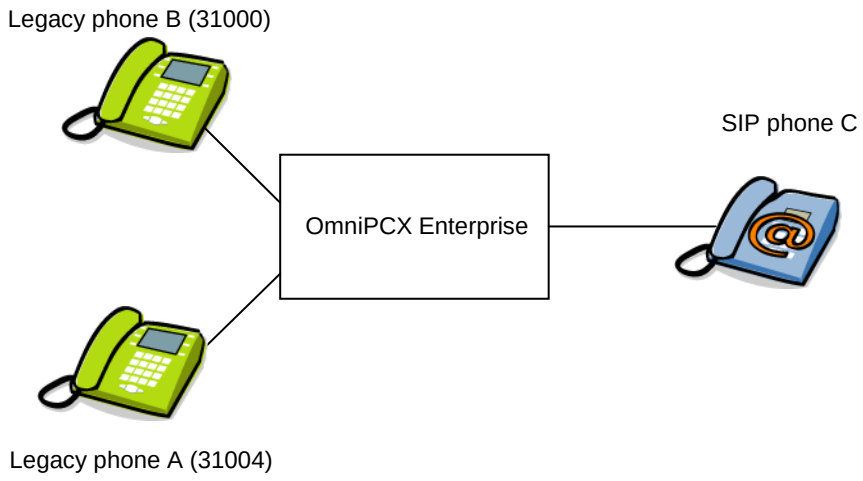
- CSIPStateByeWait200 -> CSIPStateIdle
  - Changing state from the 200 Ok of the BYE to the next INVITE (next call)

11.10 Main call flows explanation

11.10.1 Forwards

The OXE is able to manage different types of fowards, when an equipment is doing a forward to a SIP equipment, according to this forward type, the behavior and the SIP messages are different.

Topology for explanation:



11.10.1.1 Phone A calls B, and B is in direct foward to C.

In this type of call the OXE is sending an INVITE to C (for all types of fowards) , here the different type of INVITEs send according to the declaration of the SIP equipment on OXE:

- C is declared as SIP extension:

```
----------------------utf8-----------------------
INVITE sip:31026@172.27.141.210:27836;rinstance=e26a48b411982396 SIP/2.0
Allow: INVITE, ACK, CANCEL, BYE, PRACK, NOTIFY, REFER, SUBSCRIBE, OPTIONS, UPDATE, INFO
Supported: histinfo,replaces,timer,path
User-Agent: OmniPCX Enterprise R10.0 j1.410.45
Session-Expires: 1800;refresher=uac
Min-SE: 900
Content-Type: application/sdp
To: "IPtouch 172.27.141" <sip:31000@oxe-ov.alcatel.fr;user=phone>
From: "IPtouch 172.27.142.64" <sip:31004@oxe-ov.alcatel.fr;user=phone>;tag=fc0ad7be3c9267a849d2
789c08cf26d3
Contact: <sip:31004@oxe-ov.alcatel.fr;transport=UDP>
Call-ID: 3b392056e4729fbd0734266cac4106bf@172.27.141.151
CSeq: 960429378 INVITE
Via: SIP/2.0/UDP 172.27.141.151;branch=z9hG4bKc2893fd8925d9aa6704859e3fb78877a
Max-Forwards: 70
Content-Length: 240
```

In that case, the important information it is the “TO” field containing the directory number of the user forwarded to the SIP extension (31000 in that case), no mor information to indicate that the call is forwarded.

- C is declared as SIP device or an external SIP gateway:

```
----------------------utf8-----------------------
INVITE sip:31026@172.27.141.210:17680;rinstance=3e53f382fc6e4647 SIP/2.0
Allow: INVITE, ACK, CANCEL, BYE, PRACK, NOTIFY, REFER, SUBSCRIBE, OPTIONS, UPDATE, INFO
Supported: histinfo,replaces,timer,path
User-Agent: OmniPCX Enterprise R10.0 j1.410.45
Session-Expires: 1800;refresher=uac
Min-SE: 900
History-Info: <sip:31000@oxe-
ov.alcatel.fr?reason=SIP%3bcause%3d302%3btext%3d%22Moved%20Temporarily%22>;index=1,<sip:31026@oxe-o
v>;index=1.1
Content-Type: application/sdp
To: <sip:31000@oxe-ov.alcatel.fr;user=phone>
From: "IPtouch 172.27.1" <sip:31004@oxe-ov.alcatel.fr;user=phone>;tag=4200fe39737a85684b86a11b9078a0c6
Contact: <sip:31004@oxe-ov.alcatel.fr;transport=UDP>
Call-ID: bc76895c290eb936cff16ebd013b711f@172.27.141.151
CSeq: 7963653 INVITE
Via: SIP/2.0/UDP 172.27.141.151;branch=z9hG4bKcbbca67dd61c80b972173fb10c31900e
Max-Forwards: 70
Content-Length: 240
```

In that case, the important information is the “TO” field containing the directory number of the user forwarded to the SIP extension (31000 in that case), and the field “History-Info”, this information is present in case of forward and if it is managed on the OXE side for the SIP Trunk Group associated to the SIP gateway.
The “History-Info” contains the directory number of the set forwarded, the reason of forward and the destination of the forward.
The “History-Info” can be changed by “Diversion” for external SIP gateways by management.

The “History-Info” is not validated for SIP extension.

11.10.1.2 Phone A calls C, and C is forwarded to B.

```
----------------------utf8-----------------------
SIP/2.0 302 Moved Temporarily
Via: SIP/2.0/UDP 172.27.141.151;branch=z9hG4bK9e0dfb2b8f49bd46aaf944cee38cc455
Contact: <sip:31000@oxe-ov.alcatel.fr>
To: "SIP Phone"<sip:31026@oxe-ov.alcatel.fr;user=phone>;tag=16325b19
From: "IPtouch 172.27.142.64"<sip:31004@oxe-ov.alcatel.fr;user=phone>;tag=119145146a704a4541de9
Call-ID: e84e177897e67ca4977e2bb7aec3f444@172.27.141.151
CSeq: 879482083 INVITE
User-Agent: SIP Phone
Content-Length: 0

-------------------------------------------------
```

Most of the time the SIP equipment returns a 302 message to inform the proxy that the call is fowarded. This message is immediat or after a delay according to the type of forward.
If the SIP equipment is a proxy, it is able to keep the call, in that case, 2 SIP legs are opened, one from the OXE to the proxy, the second one from the proxy to the forwarded destination.

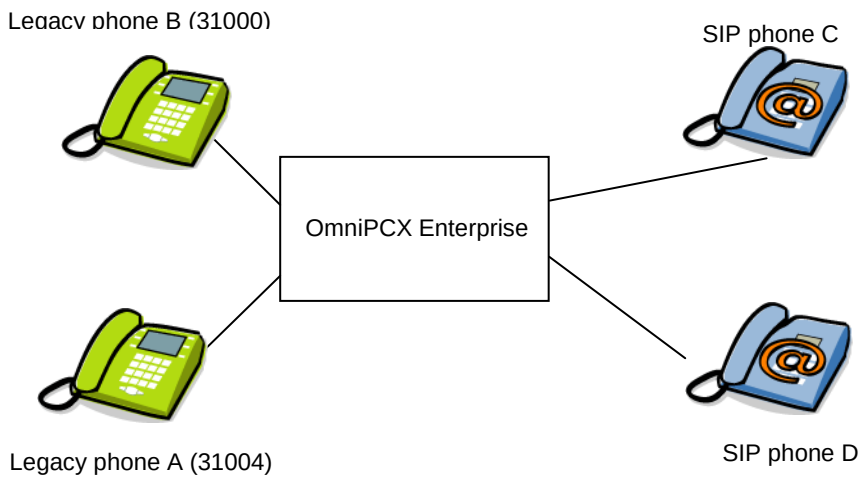
If the SIP equipment is declared as a SIP extension, from this SIP equipment the forwarding prefixes can be used, in that case no INVITE will be send to the SIP equipment, because the Call Handling knows that this user is forwarded.

11.10.2 Transfer

To make a transfer, the OXE can use (receive and accept) different ways according to the call context:

- The REFER without Replaces
- The REFER with Replaces
- The REINVITE with Replaces

<u>Topology for explanation:</u>



11.10.2.1 Use of REFER without replaces.

C calls A and C makes a transfer to B

- C sends a REFER to the SIPMOTOR

```
----------------------utf8-----------------------
REFER sip:oxe-ov.alcatel.fr SIP/2.0
Via: SIP/2.0/UDP 172.27.141.210:63016;branch=z9hG4bK-d87543-5c3865307254f255-1--d87543-;rport
Max-Forwards: 70
Contact: <sip:31026@172.27.141.210:63016>
To: "31004"<sip:31004@oxe-ov.alcatel.fr>;tag=171c87e6f9b80ed5f6819b411a72505c
From: "31026"<sip:31026@oxe-ov.alcatel.fr>;tag=15672359
Call-ID: ODFlNGNmY2JjNDgyOGEwNDRmYjhhY2NjODAxM2U2NWE.
CSeq: 3 REFER
User-Agent: SIP Phone
Refer-To: <sip:31000@oxe-ov.alcatel.fr>
Referred-By: <sip:31026@172.27.141.210:63016>
Content-Length: 0

-------------------------------------------------
```

On this REFER, the next information are present:

- "Refer-To" contains the the directory number of the transfer destination.
- "Referred-By" contains the directory number of the user doing the transfer.

- The SIPMOTOR sends a 202 Accepted to C

```
Mon Jun 25 12:04:30 2012 SEND MESSAGE TO NETWORK (172.27.141.210:63016 [UDP]) (BUFF LEN = 665)
----------------------utf8-----------------------
SIP/2.0 202 Accepted
Allow: INVITE, ACK, CANCEL, BYE, PRACK, NOTIFY, REFER, SUBSCRIBE, OPTIONS, UPDATE
Contact: sip:oxe-ov.alcatel.fr
Supported: replaces,timer,path,100rel
User-Agent: OmniPCX Enterprise R10.0 j1.410.45
P-Asserted-Identity: "IPtouch 172.27.142.64" <sip:31004@oxe-ov.alcatel.fr;user=phone>
To: "31004" <sip:31004@oxe-ov.alcatel.fr>;tag=171c87e6f9b80ed5f6819b411a72505c
From: "31026" <sip:31026@oxe-ov.alcatel.fr>;tag=15672359
Call-ID: ODFlNGNmY2JjNDgyOGEwNDRmYjhhY2NjODAxM2U2NWE.
CSeq: 3 REFER
Via: SIP/2.0/UDP 172.27.141.210:63016;received=172.27.141.210;branch=z9hG4bK-d87543-5c386530725
4f255-1--d87543-;rport=63016
Content-Length: 0
-------------------------------------------------
```

The 202 Accepted is send to accept the REFER, but the transfer is not yet done.

- The SIPMOTOR sends a NOTIFY to C

```
----------------------utf8-----------------------
NOTIFY sip:31026@172.27.141.210:63016 SIP/2.0
Content-Type: message/sipfrag
Contact: sip:oxe-ov.alcatel.fr
Supported: replaces,timer,path,100rel
User-Agent: OmniPCX Enterprise R10.0 j1.410.45
Event: refer
Subscription-State: terminated;reason=noresource
To: sip:31026@oxe-ov.alcatel.fr;tag=15672359
From: "31004" <sip:31004@oxe-ov.alcatel.fr>;tag=171c87e6f9b80ed5f6819b411a72505c
Call-ID: ODFlNGNmY2JjNDgyOGEwNDRmYjhhY2NjODAxM2U2NWE.
CSeq: 1644340323 NOTIFY
Via: SIP/2.0/UDP 172.27.141.151;branch=z9hG4bK65961cae897ba970a6b559776cd2cf88
```

The NOTIFY corresponds to the final state of the transfer, here the NOTIFY has “200 Ok” at the end of the message, in that case the transfer has be done by the OXE.
If the on NOTIFY, the information is 503 Unavailable, in that case, the transfer has failed. Some other information can be present (488, 486, etc...) according to the failed cause.

- C replies to this NOTIFY

```
----------------------utf8-----------------------
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.27.141.151;branch=z9hG4bK65961cae897ba970a6b559776cd2cf88
Contact: <sip:31026@172.27.141.210:63016>
To: <sip:31026@oxe-ov.alcatel.fr>;tag=15672359
From: "31004"<sip:31004@oxe-ov.alcatel.fr>;tag=171c87e6f9b80ed5f6819b411a72505c
Call-ID: ODFlNGNmY2JjNDgyOGEwNDRmYjhhY2NjODAxM2U2NWE.
CSeq: 1644340323 NOTIFY
User-Agent: SIP Phone
Content-Length: 0

------------------------------------------------
```

11.10.2.2 Use of REFER with replaces.

C calls A and C calls D and makes a transfer

- C sends a REFER to the SIPMOTOR to replace an existing dialog

```
----------------------utf8-----------------------
REFER sip:oxe-ov.alcatel.fr SIP/2.0
Via: SIP/2.0/UDP 172.27.141.210:63016;branch=z9hG4bK-d87543-d60505761b7d746d-1--d87543-;rport
Max-Forwards: 70
Contact: <sip:31026@172.27.141.210:63016>
To: "31004"<sip:31004@oxe-ov.alcatel.fr>;tag=0219e846e66c868f72a9dbdfa8e58e2a
From: "31026"<sip:31026@oxe-ov.alcatel.fr>;tag=9c131c4f
Call-ID: ZTBjODRhNzFhYTY3ZGNiMjI4N2FjZDQzNTI2MjA2YjE.
CSeq: 7 REFER
User-Agent: SIP Phone
Refer-To: "31023"<sip:31023@oxe-ov.alcatel.fr?Replaces=YTI4MmJhZjcyMDAyYmYyODI2ZmU0NmE5MWVhMGU2MDc.%3Bto-
tag%3D053621a0570c23654c20fb10154dd7f5%3Bfrom-tag%3D7728f179>
Referred-By: <sip:31026@172.27.141.210:63016>
Content-Length: 0
------------------------------------------------
```

On this call flow there are three legs:

- Leg1 corresponds to the call from C to A
- Leg2 corresponds to the call from C to D for the direction C to SIPMOTOR
- Leg3 corresponds to the call from C to D for the direction SIPMOTOR to D

On this REFER, the next information are present:

- "Refer-To" contains the directory number of the transfer destination with a "Replaces" corresponding to the leg to replace (leg2)
- "Referred-By" contains the directory number of the user doing the transfer.

At the end of the transfer the leg1 is closed by C and leg2 is closed by the SIPMOTOR, it is remaining only the leg3 from the A to D.

11.10.2.3 Use of REINVITE with replaces.

C calls A and C calls D and C makes a transfer

- C sends a REINVITE to the SIPMOTOR to replace an existing dialog

```
----------------------utf8-----------------------
INVITE sip:oxe-ov.alcatel.fr SIP/2.0
Via: SIP/2.0/UDP 172.27.141.210:63016;branch=z9hG4bK-d87543-71672411fa2ca01c-1--d87543-;rport
Max-Forwards: 70
Contact: <sip:31026@172.27.141.210:63016>
To: "31004"<sip:31004@oxe-ov.alcatel.fr>;tag=0219e846e66c868f72a9dbdfa8e58e2a
From: "31026"<sip:31026@oxe-ov.alcatel.fr>;tag=9c131c4f
Call-ID: ZTBjODRhNzFhYTY3ZGNiMjI4N2FjZDQzNTI2MjA2YjE.
CSeq: 6 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO
Referred-By: <sip:31026@172.27.141.210:63016>
Replaces=YTI4MmJhZjcyMDAyYmYyODI2ZmU0NmE5MWVhMGU2MDc.%3Bto-tag%3D053621a0570c23654c20fb10154dd7f5%3Bfrom-
tag%3D7728f179>
Content-Type: application/sdp
User-Agent: SIP Phone
Content-Length: 256
```

The principle is the same than a REFER with replaces, but it is a REINVITE message

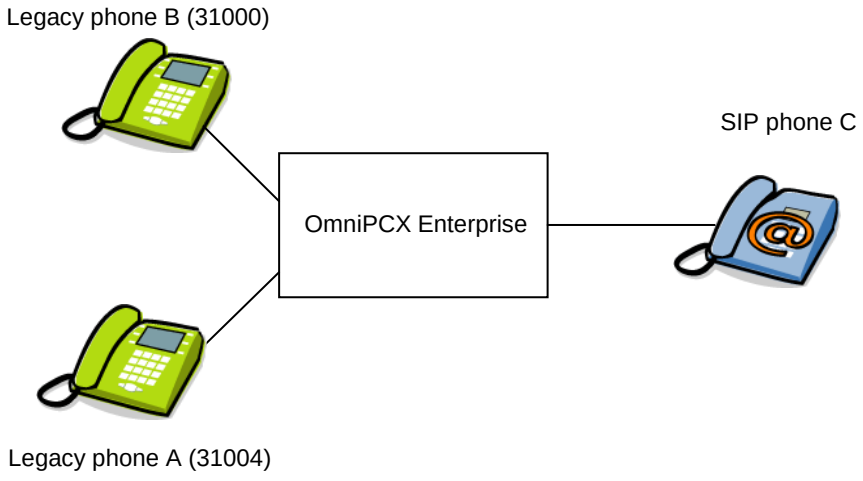
On this REINVITE, the next information are present:

- "Referred-By" contains the directory number of the user doing the transfer.
- "Replaces" contains the the directory number of the transfer destination with a "Replaces" corresponding to the leg to replace (leg2).

11.10.3 UPDATE on Early Media

In some calls scenarios, the OXE will send or receive an UPDATE on Early Media (before dialog opened) to change the SDP.

Topology for explanation:



- Phone A calls B, B calls C and makes a blind transfert to C.

During the RINGING phase, the OXE will send to the C a UPDATE (after sending the 180 RINGING) to C. To send the UPDATE, the OXE needs before to send a PRACK, to make a Pre-Acknowledgment and receive a 200ok for this PRACK, after this, the OXE will be able to send an UPDATE.

- To send a PRACK the OXE needs a “Require: 100rel” on the 18X answer received:

```
Mon Jun 11 15:01:38 2012 RECEIVE MESSAGE FROM NETWORK (172.27.143.186:5060 [UDP])
----------------------utf8-----------------------
SIP/2.0 180 Ringing
Allow: INVITE, ACK, CANCEL, BYE, PRACK, NOTIFY, SUBSCRIBE, OPTIONS, UPDATE
Contact: sip:172.27.143.186
Require: 100rel
User-Agent: SIP Phone
To: <sip:31006@172.27.143.186;user=phone>;tag=d7758dbc7f49c9521d28e60ef312ab04
From: "IPtouch 172.27.1" <sip:31000@oxe-ov.alcatel.fr;user=phone>;tag=0c835efa2e1bf86a90d0016a
Call-ID: d626cd71ab0eab5c0499c46fd5324a91@172.27.141.151
CSeq: 679245852 INVITE
Via: SIP/2.0/UDP 172.27.141.151;branch=z9hG4bK61c571ebc4b1f5e5ff9e122e7e8b4a06
RSeq: 1131790336
Content-Length: 0
-------------------------------------------------
```

- After receiving this “Require: 100rel”, the OXE generates the PRACK

```
Mon Jun 11 15:01:38 2012 SEND MESSAGE TO NETWORK (172.27.143.186:5060 [UDP]) (BUFF LEN = 514)
----------------------utf8-----------------------
PRACK sip:172.27.143.186 SIP/2.0
Supported: replaces,timer,path
User-Agent: OmniPCX Enterprise R10.0 j1.410.45
RAck: 1131790336 679245852 INVITE
To: <sip:32006@172.27.143.186;user=phone>;tag=d7758dbc7f49c9521d28e60ef312ab04
From: <sip:31000@oxe-ov.alcatel.fr;user=phone>;tag=0c835efa2e1bf86a90d0016a0389c18e
Call-ID: d626cd71ab0eab5c0499c46fd5324a91@172.27.141.151
CSeq: 679245853 PRACK
Via: SIP/2.0/UDP 172.27.141.151;branch=z9hG4bK8b757b21da861556184ff74e5f5aaca7
Max-Forwards: 70
Content-Length: 0
-------------------------------------------------
```

- The OXE receives the 200ok of the PRACK

```
Mon Jun 11 15:01:38 2012 RECEIVE MESSAGE FROM NETWORK (172.27.143.186:5060 [UDP])
----------------------utf8-----------------------
SIP/2.0 200 OK
Allow: INVITE, ACK, CANCEL, BYE, PRACK, NOTIFY, SUBSCRIBE, OPTIONS, UPDATE, INFO
Supported: timer,path,100rel
User-Agent: OmniPCX Enterprise R10.0 j1.410.45
To: <sip:32006@172.27.143.186;user=phone>;tag=d7758dbc7f49c9521d28e60ef312ab04
From: <sip:31000@oxe-ov.alcatel.fr;user=phone>;tag=0c835efa2e1bf86a90d0016a0389c18e
Call-ID: d626cd71ab0eab5c0499c46fd5324a91@172.27.141.151
CSeq: 679245853 PRACK
Via: SIP/2.0/UDP 172.27.141.151;branch=z9hG4bK8b757b21da861556184ff74e5f5aaca7
Content-Length: 0
-------------------------------------------------
```

- The OXE sends the UPDATE to change the SDP.

```
Mon Jun 11 15:01:38 2012 SEND MESSAGE TO NETWORK (172.27.143.186:5060 [UDP]) (BUFF LEN = 895)
----------------------utf8-----------------------
UPDATE sip:172.27.143.186 SIP/2.0
Supported: replaces,timer,path
User-Agent: OmniPCX Enterprise R10.0 j1.410.45
RAck: 1131790336 679245852 INVITE
To: <sip:32006@172.27.143.186;user=phone>;tag=d7758dbc7f49c9521d28e60ef312ab04
From: <sip:31000@oxe-ov.alcatel.fr;user=phone>;tag=0c835efa2e1bf86a90d0016a0389c18e
Call-ID: d626cd71ab0eab5c0499c46fd5324a91@172.27.141.151
CSeq: 679245852 UPDATE
Via: SIP/2.0/UDP 172.27.141.151;branch=z9hG4bK8b757b21da861556184ff74e5f5aaca7
Max-Forwards: 70
Content-Type: application/sdp
Content-Length: 291

v=0
o=OXE 1339422663 1339422663 IN IP4 172.27.141.151
s=abs
c=IN IP4 172.27.142.64
t=0 0
m=audio 32514 RTP/AVP 18 97
a=sendrecv
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=ptime:20
a=maxptime:40
a=rtpmap:97 telephone-event/8000
------------------------------------------------
```

The UAS, receiving this UPDATE, is able to use the connection point for the RTP flow

11.11 Configuration issues

Most of the SIP issues are linked to a bad management.

When you connect a SIP equipment, it is mandatory to check if this equipment is tested and validated by Alcatel-Lucent

- The SIP equipments like faxs, sets, etc… are validated via the AAPP. The Configuration procedures are available on BPWS.
- The SIP providers are tested themselves the OXE, so if you want to connect one SIP provider, check if this provider has done the interopability test. All the configuration procedures are given by the providers and not by Alcatel-Lucent.

If a SIP equipment connected is not validated by Alcatel-Lucent, no support will be provided.

11.11.1 SIP configuration rule

- General Parameters
  - DPNSS prefix (necessary for optimisation on call forward).
  - System codec (G729, G723).
  - Support of multi-algo should be set to false.

- Netadmin
  - Use of specific characters (& _ $ ...) is not allowed for the nodename.

- Activate internal name resolver in spatial redundancy topologies.

- Local SIP gateway
  - The local SIP gateway is managed when the SIP Trunk group and the SIP Subnetwork are managed (minimum of configuration to do).
    - Alcatel-Lucent recommends to use an ABCF SIP Trunk Group on the local SIP gateway
    - The network number is a free one, must not used by another application (ABCF network, Hybrid links, VPN hop, etc…).
    - This network number is the same than the one managed on the SIP ABCF Trunk Group linked to this local SIP gateway.

- External SIP Gateway
  - The external SIP gateway can use the same Trunk Group (TG) than the local SIP gateway.
  - The external SIP gateway can use another Trunk Group.
    - If it is an ABCF TG, the network number manage for this TG is different than the one used on the TG for the local SIP gateway.
    - If it is an ISDN TG, let the OXE manages the network number by itself, in fact the configuration is the same than a real ISDN T2/T1.
  - If the external SIP gateway is with an ISDN SIP TG, **only ARS must be used**, no network or routing numbers.
  - If the external SIP gateway is with an ABCF SIP TG, network or routing numbers can be used without restrictions, if the ARS is used, the OXE must not receive REFER (or REINVITE with replaces) or 30X messages on this external SIP gateway (ARS limitation).

- SIP Trunk group
  - ABCF TG: no restrictions about SIP messages.
  - ISDN SIP TG: no REFER (or REINVITE with Replaces) or 30X messages will be sent and received.

- SIP Proxy
  - By default, the SIP proxy is set with “SIP Digest” for the Minimal authentication method, but there is no Realm managed, so it is necessary to disable the authentication (SIP None) or to manage a Realm.

- In case of SSH management, the SIP equiments must be managed as SIP gateway (choice 1).

11.11.2 SIP alarms generated on OXE

On the OXE SIP incidents are generated on Call Handling side, thes incidents are linked to a SIP alarm (files under /tmpd), here an example of SIP alarm generated:

- Alarm due to Subscriptions:

```
> 02/07/12 - 15:39:35       Warning alarm
       37F6 [CResponse::checkResponseFields] unknown header is not applicable for 202/SUBSCRIBE responses

> 02/07/12 - 15:39:35       Minor alarm
       [CSubscriptionState::receiveSubscribeMessage] Call: 28844ea68ff53075 eqt: -1 SUBSCRIPTION_STATE failed to
emit a Successful message.
```

In that situation, the OXE receives a “SUBSCRIBE” message, but is not able to answer it, because the purpose of this “SUBSCRIBE” message is unknow by the OXE.

When this type of alarms are present on the OXE, remove the Subscription on the remote SIP equipment, to avoid the Alarm.

When lot of alarms like this one are generated on OXE, they can cause a “crash” of the SIPMOTOR.

- Alarm due to bad SIP call context not copied on Stand-By CPU:

```
> 02/07/12 - 15:39:35       Warning alarm
       37F6 [receiveInviteMessage] StandByCallCreation failed !.
```

On the trace, these information are present:

```
1309553189 -> [CDuplicateCall::create_duplication_data_struct] _ViaSet size 218.
1309553189 -> [CDuplicateCall::create_duplication_data_struct] Via is bigger than uiCAlcStrStaticGrow:192 -
RealSize:218.
1309553189 -> ALARM: [receiveInviteMessage] StandByCallCreation failed !.
```

In that situation, on the INVITE received, the VIA header is too long for the OXE and the it is not able to send the SIP “context” to the stand by CPU. The call is established, but in case of bascul, this will not be known by the new main CPU.

- Alarm to emit an INVITE message:

```
> 02/07/12 - 15:39:35       Minor alarm
       [receiveInviteEvent] Call:  eqt: 30311 INITIAL_STATE failed to emit an Invite message.
```

When the Information is “receiveInviteEvent”, the Call Handling is sending an INVITE to the SIPMOTOR, but due to a lack of ressources or licenses the INVITE cannot be emitted by the SIPMOTOR.

```
> 02/07/12 - 15:39:35       Minor alarm
       [receiveInviteMessage] failed to emit an Invite event.
```

When the Information is “receiveInviteMessage”, the SIPMOTOR has recieved an INVITE but due to a lack of ressources (channels on SIP Trunk Group, CAC, compressors, ...) or licenses, the SIPMOTOR cannot send the INVITE to the Call Handling.

- Alarm due to a request not for the SIP proxy of the OXE:

```
> 06/05/12 - 21:56:44       Warning alarm
       [CIOCom::receiveResponse] Received response is not for this entity
```

This alarm means that the SIPMOTOR receives a SIP resuqets not for it, and is not able to rout it to another SIP equipment. Necessary to make a SIPMOTOR traces to get the IP address of this SIP equipment.

- Alarm to emit a SIP message MESSAGE:

```
> 06/05/12 - 22:14:46        Minor alarm
       [receiveMessageEvent] Call:  eqt: 2862 INITIAL_STATE failed to emit an instant message.
```

The SIPMOTOR is not able to send a SIP message MESSAGE to a SIP extension, remove the fact to send this message on the SIP extension phone cos.

- Alarm to emit a SIP message CANCEL:

```
> 03/08/12 - 09:31:11   Minor alarm
        [receiveCancelEvent] Call: 112c581b1c96acc94a45f53f96e5591a@172.27.141.151 eqt: 2175 COMPLETED_STATE
failed to emit a Cancel message.
```

The SIPMOTOR generates this alarm because it is not able to send a CANCEL message, because the dialog is already opened. The Call Handling asks the SIPMOTOR to send a CANCEL, but the 200ok for this INVITE transaction is already arrived.

- Alarm to emit a SIP message ACK:

```
> 02/24/12 - 16:31:42   Minor alarm
        [receiveAckEvent] Call: c40c7cd3a74a5bdf7457bc28586650f2@172.27.141.151 eqt: 2175 TERMINATED_STATE failed
to emit an Ack message.
```

The SIPMOTOR generates this alarm because it is not able to ACK an INVITE transaction, because the transaction is already terminated. Need to open a SR for analyse.

11.11.3 Common SIP issues

This part is used to explain the general issues possible on the OXE, not for a specific equipment

11.11.3.1 SIPMOTOR

- Issue 1: No SIPMOTOR processes are running
  - Symptom: With the command ps -edf | grep sipmotor, no processes are present
  - Explanation: This is due to a bad configuration of the SIP on your OXE, for instance, the SIP Trunk group manage on the local SIP gateway is not a SIP Trunk Group.
  - Solution: Manage the good configuration, and after a restart of the CPU is mandatory.
- Issue 2: Only 2 SIPMOTOR processes are running
  - Symptom: With the command ps -edf | grep sipmotor, only 2 SIPMOTOR processes are present

- Explanation: When a modification is done on the SIP Trunk Group associated to the local SIP gateway, for instance to replace Mini SIP Trunk group by a SIP Trunk group, the OXE needs do resize the memory space due to this modification (often after the first management of the local SIP gateway)

- Solution: A restart of the CPU is mandatory

- Issue 3: SIPMOTOR in degraded mode

  - Symptom: SIPMOTOR is rejecting all the call by a 503 message, and with the tool "sipdump", the status of the SIPMOTOR is in "degraded" mode

  - Explanation: This a protection for the SIPMOTOR, when there are too much SIP "instance" in the SIPMOTOR, the SIPMOTOR switches in degraded mode to protect itself. When it has this status, all the incoming SIP requests are rejected by a 503. This mechanism avoids the application from being overwhelmed by the traffic.

  - Solution: nothing can be done, the SIPMOTOR will disable this mode automaticaly due to some internal timers and thresholds.

- Issue 4: Losing all the SIP call contexts

  - Symptom: If a restart of the SIPMOTOR is performed, all the SIP call contexts are lost

  - Explanation: The restart of the SIPMOTOR provides the loss of all the SIP contexts, that means, if you have SIP calls established, the RTP flow is maintained, for the SIP point view the call is n ot anymore present, that means, if the SIPMOTOR receives a BYE for a call, the BYE will be answered by a "481 Call/Transaction Does Not Exist", but the call will be stopped. Also if you use the session timer (time to check if the call is still up for the SIP point of view) the call will be cut by the OXE because, the context is unknown by the SIPMOTOR

  - Solution: This is a normal behaviour if the restart is done manually. If the SIPMOTOR restart automaticaly, in case a SR must be opened for analyse.

- Issue 5: SIPMOTOR memory leak.

  - Symptom: The SIPMOTOR is using more and more memory space.

  - Explanation: When the SIP is managed on the OXE, the SIPMOTOR processes are using memory space, when the traffic is going up, the memory space used is increasing, when the traffic is going down, the memory space used is decreasing. Now, when the traffic is going down, the memory space used doesn't decrease correctly, and if day after day, when there is no traffic, the memory used is more higher, the SIPMOTOR will crash soon. In such case, the SIPMOTOR has problems to "delete" the SIP contexts from its memory, and after accumullation of the SIP contexts not deleted, the SIPMOTOR cannot work properly, and crash.

  - Action to do:

    - Check if the configuration of the OXE respects the Alcatel-Lucent recommendations.
    - Check if the REGISTER messages received on SIPMOTOR are not too much, the registration of a SIP equipments must not be used as a "keep alive".
    - Check if the SIPMOTOR doesn't receive SIP messages not for it.

- Check if the SIPMOTOR doesn't receive SUBSCRIBE messages not used by OXE.

- Solution: A restart of the SIPMOTOR can be done and due to this, all the SIP contexts are deleted. The problem will be solved but only for a time, if the root cause is not found, the problem will be back again. Open a SR for analyse.

11.11.3.2 Call failure

- Issue 1: Incoming SIP calls are cut by the OXE after 32 seconds:

  - Symptom: Incoming SIP calls are cut by the OXE after ~3 secondes (or 32 seconds in case of SIP extension) and the 200ok from OXE is never ACK by the external SIP equipment.

  - Explanation: If the system is in spatial redundancy, check if the FQDN of the OXE is used by the external SIP equipement, in fact on the "Contact", the FQDN is added by the OXE, this FQDN is unknown by the SIP equipment (because it is using the IP address), and it doesn't answer to this 200ok, the OXE send several times the 200ok and the OXE cuts the call because no ACK for this call.

  - Solution: The remote SIP equipment must use the FQDN of the OXE. Since the R10, a parameter is present on the external SIP gateway only "Contact with IP address" used to put the IP address of the main CPU instead of the FQDN in the Contact header.

- Issue 2: Calls are not anymore possible from a SIP equipment:
  - Symptom: The SIP calls are not possible thru an external SIP gateway in high traffic.
  - Explanation: Check if the IP address managed on the external SIP gateway is put in quarantine (in sipalarm files)
  - Solution: Manage the IP address on the trusted SIP IP addresses. A restart of the SIPMOTOR is mandatory after management.
- Issue 3: SIP calls are rejected with a 502:
  - Symptom: When a SIP call, using an ABCF SIP Trunk Group, to an external number is not possible (thru a carrier for instance) and rejected most of the time by a 502 Bad Gateway. Internal calls are ok and incoming calls also ok for this SIP equipment.
  - Explanation: When the message 502 is reponded to a SIP request, the problem is due to the management, that means, the information on the SIP request are not good for the call in progress. In that case, the call is done from an ABCF SIP Trunk Group to an external called party, the call is rejected because the DID transcoding is set to “True” on the ABCF SIP Trunk Group
  - Solution: Set the “DID transcoding” of the SIP ABCF Trunk group to false (mandatory).
- Issue 4: SIP calls are rejected with a 488 Not Acceptable here:
  - Symptom: When a SIP call is rejected by 488 SIP message,
  - Explanation: When a SIP call arrives on the OXE, the Call Handling checks if the SDP received is compatible for this call, if it is not the case, the Call Handling asks the SIPMOTOR to send a response 488 for this request
  - Solution: Manage the SDP of the SIP equipment to be compatible with the configuration of the OXE or on the other way
- Issue 5: SIP calls are rejected with different reasons:
  - Symptom: When a SIP call is rejected by 488, 502, 404, etc...
  - Explanation: When a SIP call arrives on the OXE, this call is automatically rejected by OXE, but the reason can be different, even if the scenario of the call is the same. The SIP is linked to the shelf 19 associated to the CPUs, so if the CPUs are not belonging to the IP domain 0, the virtual INTIP boards of the shelf 19 doesn’t belong to the IP domain 0, and the SIP is affected by this configuration.
  - Solution: Manage CPUs IP addresses on the IP domain 0, this mandatory in case of SIP.
- Issue 6: SIP calls are rejected with 403 No license available:
  - Symptom: When a SIP call is rejected by 403 No license available.

- Explanation: When a SIP call is done, a license is used for this call. In case of incoming call, if no more license is available, the OXE rejects the call by a 403 No licenses available. The problem can be only the number bought by the customer, it is no enough according to the number of simultaneous SIP calls, or some SIP call contexts are blocked on the SIPMOTOR.

- Action to do:

    - When no more SIP calls, restart the SIPMOTOR.
    - Run the SIPMOTOR traces:
      >motortrace 3 (or 6)
      >traced -l /tmpd/traced -s 10000000 -f 50 -d &
    - Keep the trace running until the issue is present.
    - When the issue is present, run “sipdump” and make the choice 1 and 4 every minutes during 5/10 minutes.
    - Stop the traces
    - When no more SIP calls are present on OXE, run the next trace (do not restart the SIPMOTOR!!!):
      >motortrace 3 (or 6)
      >traced >/tmpd/trace_sip.log and make one call and stop it.

  On the file “trace_sip.log”, search for “nb available licenses=”.

- Solution: If the number of licenses is the number of the licenses bought on OXE, there is no issue, the solution is to buy more licenses. If the number is less than the number bought, open a SR and provide the traces files and the Infocollect of the site.

11.11.4 SIP Device issues

An important thing to remenber about SIP device, it is that all the calls are linked to the SIP Trunk Group associated to the local SIP Gateway, so if you manage a SIP ABCF Trunk Group or an ISDN SIP Trunk Group, the behaviour will be different.

- Issue 1: Forward on no reply doesn’t work when the destination is a SIP device:

    - Symptom: It is not possible to make a forward on no reply (on an IPtouch for instance) when the destination is a SIP device, ok for immediat foward.

    - Explanation: The SIP device behavior is linked to the SIP Trunk group associated to the local SIP gateway, if you use an ISDN SIP TG, or an ABCF SIP TG, the behaviour will be different. The SIP Trunk Group used on the local SIP gateway is a SIP ISDN TG.

- Solution: Change the SIP Trung Group managed on the local SIP gateway from SIP ISDN TG to SIP ABCF TG, restart of the SIPMOTOR is mandatory.

- Issue 2: Afer a while, all SIP phones registrations and subscriptions are impossible

  - Symptom: More 1000 SIP Devices losses their registration. Only a double bascul of PBX resolves this issue

  - Explanation: As there are more 1000 SIP devices which register/subscribe at the same moment, there are too much of traffic to be managed by the PBX and resources on SIPMOTOR are blocked. Around 45000 Subscription and Registration been handled in 3 hours time. This is really a big number,Oxe is dealing with. Solution shoud be to stop some of the unwanted Subscribe messages. And increase the subscriptions and registration timers on SIP Devices. Unwanted subscriptions meant here was even though voice mail was not configured for a phone set, subscription value was configured, this should be 0.

Example of Registration too brief:

```
Sun Sep 30 06:53:09 2012 RECEIVE MESSAGE FROM NETWORK (172.30.125.75:5060 [UDP])
----------------------utf8-----------------------
REGISTER sip:172.30.127.2:5060 SIP/2.0
Expires: 60

1348980789 -> Sun Sep 30 06:53:09 2012 SEND MESSAGE TO NETWORK (172.30.125.75:5060 [UDP])
(BUFF LEN = 394)
----------------------utf8-----------------------
SIP/2.0 423 Registration Too Brief
Min-Expires: 1800
```

Example of sipalarm when subscription is impossible on /tmpd:

```
[CSubscriptionState::receiveSubscribeMessage] eqt: -1 SUBSCRIPTION_STATE failed to emit a Successful message.
```

Example of DHCP buffer issue on /varlog/messages:

```
Nov 7 00:01:52 sr_cpub dhcpd: send_packet: No buffer space available
Nov 7 00:01:52 sr_cpub kernel: Neighbour table overflow.
Nov 7 00:01:52 sr_cpub kernel: Neighbour table overflow.
```

  - Solutions:
    1. Increase registration and susbcriptions timers on SIP Devices from 60 secondes to 1800.
    2. Deactivate unnecessaries subscriptions sent to PBX when no services are configured on users management, example: if Voicemail is available via another application, subscription must not sent to PBX
    3. Configure a dedicated VLAN for OXE (CS, GD) and one or more VLANs for SIP Devices in order to decrease ARP requests on DHCP service

With the current Linux OS, OXE has a limitation in handling more than 1000 data equipment if it is connected in the same sub-network. So we need to have a seperate VLAN in between to handle this. OXE CS must be placed under separate subnet and the IP Phones distributed under different other subnets

11.11.5 SIP extension issues

The SIP extension is not linked to a SIP Trunk Group, it can be created without SIP management

- Issue 1: SIP fax equipment, declared as a SIP extension, doesn't work:

- Symptom: when a SIP fax equipment tries to make a call, the REINVITE for the T38 negociation is never seen

- Explanation: When a SIP fax call is done, the establishement of the call is done in two phases, opening of RTP channel then opening of a T38 channel, in case of SIP extension, the T38 is not implemented, so the second phase cannot be done, and the call is stopped

- Solution: Use of a SIP Device user instead of a SIP extension

- Issue 2: SIP extension multiline, SIP phone monoline:

  - Symptom: when a SIP extension is created, it is a multiline user, and if the SIP phone is associated is monoline, the functioning of the SIP extension can cause issue

  - Explanation: A SIP extension user, declared in “business” mode, is multiline, that means taht teh SIP phone associated must be multiline as well, if it is not the case, the call to the second line of the user is rejected by the SIP phone, and this can cause disturbances on the SIP extension behaviour (call handling side) .

  - Solution: A SIP phone associated to a SIP extension user must be multiline.

11.11.6 SIP External Gateway Issue

- Issue 1: One way calls after remote SIP equipment put on hold and retreive the call:

  - Symptom: A SIP call is done between the OXE and a remote SIP gateway, this SIP equipment puts on hold the call, the OXE equipment can hear the MOH, and when the SIP equiment retrieves it, the one way call is present.

  - Explanation: When the SIP external gateway put on hold, it is sending a REINVITE with a “Black Hole” (c=0.0.0.0 on SDP) or an “INACTIVE” to stop the RTP flow, before to send a new REINVITE with a SDP for MOH. When a new REINVITE is sent to get back the converstaion, the OXE is not able to connect the RTP flow to the SDP given on this REINVITE.

  - Solution: On the external SIP gateway, put True for the parameter “Ignore inactive/black hole”, in that case, the OXE will not take into account the “Black Hole” or the “INACTIVE”.

- Issue 2: One way call in case of incoming/outgoing calls:

  - Symptom: An incoming or an outgoing calls are well established, but no speech send by OXE

  - Explanation: The problem has been seen after an upgrade from a version lower to I160516c to a R10. On the traces taken, the OXE is not getting SDP or, INVITE or 200ok. The problem was about the parameter “Routing Application”, this parameter is used for the feature “Force_on_NET”, in case of incoming call to the OXE, this call is not for an equipment connected to the OXE, but for an external user (mobile phone for instance), so for such call, the OXE doesn’t need to reserve ressources on its side. This parameter has been designed for that.

  - Solution: Put False for this parameter if it set to True.

11.12 Use case

11.12.1 Outgoing Call – Cancel sent by OXE after 180 w SDP

Symptom: SIP ISDN Outgoing call are cancelled by OXE after 180 Ringing SDP (G711) reception.

Diagnosis: - Check if CS's IP Address is configured on IP Domain 0.
- Check extra domain codec where caller is located

Solution: As only G711 codec is available for Outgoing calls (`IP Compression Type + G711 on TG`) and caller is located on a restricted domain (`Extra Domain Coding Algorithm + With Compression on IP Domain`), OXE cannot sends/receives media stream. Call is cancelled.

11.12.2 Telephone-event are not provided on SDP offer

Symptom: Re-INVITE sent by OXE to SIP Provider doesn't contain telephone event media on SDP offer

Solution: On SIP > SIP External Gateway, check at false parameter "To EMS".

11.12.3 Loss of communication with SIP External Voicemail

Symptom: Frequent loss of communication between external voicemail and OXE connected via SÏP trunk

Diagnosis: Check if congestion occurs with incident 5816 when you try to access to the voice mail.
Check if Voicemail IP Address is present on Trusted IP Addresses

Solution: Voicemail was put in quarantine and during one half hour all calls in direction of Voicemail were blocked

11.12.4 Impossible to let a message when routing via SIP Automated Attendant

Symptom: It is not possible to let a message on the voicemail of the called number in case of an automated attendant SIP and when the Phone Feature COS "Voicemail forwarding" is set at "Ring called set mail"

Solution: On System > Other System Param. > Spec. Customer Features Parameters > Voice Mail forwarding SIP auto att, check this parameter at true

11.12.5 When call is transfer from a Third Party Server, after few seconds, a Re-Invite is sent by OXE to reroute RTP to a GD card

Symptom: When call is established, after few seconds, OXE sends a reinvite request to redirect RTP to a GD card.

Solution: DPNSS is used on this scenario. On System > Other System Param. > External Signalling Parameters > DeActivate Path Replacement, check this parameter at true

11.12.6 Incoming call from a SIP Third Party Server is rejected by OXE with a SIP Error 488 Not Acceptable Here

Symptom: Incoming call is rejected by a SIP Error 488 Not acceptable Here

Diagnosis: Check Extra Domain Coding Algorithm concordance
Check Public Access Category

Solution:

- On IP > IP Domain > Extra Domain Coding Algorithm must be the same as third party offer
- On Categories > Access Category > Go down hierarchy > Public Access Category > Select COS 31 and give correct rights

11.12.7 Incoming call is not recognized as INTERNATIONAL

Symptom: Incoming call received on set phone indicates local call instead of international call.

Diagnosis: - Country code is not separated of received number by PBX so canonical form is not correctly set up. Canonical form is "+" country code "–" *(number). So, number should be +49–71182137821 in order to detect that is an international incoming call.

Solution: Add the country code 49 on External Country Code section Translator > External Numbering Plan > Coutry Codes:

Country code prefix : 49
Country Value + Germany

11.12.8 When we attempt to register on SIP External Gateway, OXE answers by a SIP error "482 Loop Detected"

Symptom: For each register sent to OXE, we have a SIP error "482 Loop Detected", as below REGISTER request:

```
1352974529 -> Thu Nov 15 11:15:28 2012 SEND MESSAGE TO NETWORK (172.27.139.90:5060 [UDP]) (BUFF LEN = 478)
----------utf8----------
REGISTER sip:hq2cs.labjtr.fr SIP/2.0
Supported: 100rel,path
User-Agent: OmniPCX Enterprise R10.1 j2.501.16.c
To: sip:4321@hq2cs.labjtr.fr
From: sip:4321@hq2cs.labjtr.fr;tag=a9ca34e0b0534fb9d4e0823b7b5d4eaa
Contact: <sip:4321@172.27.145.122;transport=UDP>;expires=1800
```

And error received:

```
Thu Nov 15 11:15:28 2012 RECEIVE MESSAGE FROM NETWORK (172.27.139.90:5060 [UDP])
----------utf8----------
SIP/2.0 482 Loop Detected
To: sip:4321@hq2cs.labjtr.fr
From: sip:4321@hq2cs.labjtr.fr;tag=a9ca34e0b0534fb9d4e0823b7b5d4eaa
Call-ID: 2f9392c14ee4303329bb32a948e74e35@172.27.145.122
CSeq: 1821162596 REGISTER
Via: SIP/2.0/UDP 172.27.145.122;branch=z9hG4bK47b7d67d20268bb0c40d57c60e4c1cb9
Content-Length: 0
```

Diagnosis: Registration is done by Domain Name resolution so the sip Request-URI sip:hq2cs.labjtr.fr must be matched with machin name filled on SIP Gateway. The SIP URL of REGISTER contains the SRV/A domain name. Proxy loops that call back to itself because it does not know about itself as the SRV/A domain.

Solution: Modify the SIP Gateway in order to have the same Machin Name as SIP URL contained on REGISTER, use the command **netadmin** to do it:

Trunk Group : 35
IP Address : 172.27.139.90
Machin name : **hq2cs.labjtr.fr**
Proxy Port Number : 5060
DNS local domain name : labjtr.fr
DNS type + DNS A
First DNS IP Address : 172.27.139.88

11.12.9 When we attempt to register our SIP External Gateway with an external SIP Proxy, SIP Proxy answers by a SIP error "416 Unsupported URI Scheme"

Symptom: For each register sent to external SIP Proxy, we have a SIP error "416 Unsupported URI Scheme", as below REGISTER request:

```
1352975879 -> Thu Nov 15 11:37:56 2012 RECEIVE MESSAGE FROM NETWORK (172.27.145.122:5060 [UDP])
----------------------utf8----------------------
REGISTER sip:hq2.labjtr.fr SIP/2.0
Supported: 100rel,path
User-Agent: OmniPCX Enterprise R10.1 j2.501.16.c
To: sip:hq2.labjtr.fr
From: sip:hq2.labjtr.fr;tag=56b8ce5bd76524902b5c171f39c9bbdf
Contact: <sip:172.27.145.122;transport=UDP>;expires=1800
Call-ID: 01f55be7e5c59d21f72659fabc36878a@172.27.145.122
CSeq: 1643105352 REGISTER
Via: SIP/2.0/UDP 172.27.145.122;branch=z9hG4bKdc224f76827da20ba9390b081ef8aed0
Max-Forwards: 70
Content-Length: 0
```

And error received:

```
Thu Nov 15 11:37:56 2012 SEND MESSAGE TO NETWORK (172.27.145.122:5060 [UDP]) (BUFF LEN = 344)
----------------------utf8----------------------
SIP/2.0 416 Unsupported URI Scheme
To: sip:hq2.labjtr.fr;tag=75e766ee37e6bf967b4c84db521f8406
From: sip:hq2.labjtr.fr;tag=56b8ce5bd76524902b5c171f39c9bbdf
Call-ID: 01f55be7e5c59d21f72659fabc36878a@172.27.145.122
CSeq: 1643105352 REGISTER
Via: SIP/2.0/UDP 172.27.145.122;branch=z9hG4bKdc224f76827da20ba9390b081ef8aed0
Content-Length: 0
```

Diagnosis: Registration ID is not present on REGISTER request so SIP Proxy cannot authenticate the OXE. Configure the parameter Registration Id on SIP External Gateway

Solution: Configure the parameter Registration Id on SIP External Gateway, as well

```
1352976351 -> Thu Nov 15 11:45:50 2012 RECEIVE MESSAGE FROM NETWORK (172.27.145.122:5060 [UDP])
----------------------utf8----------------------
REGISTER sip:hq2.labjtr.fr SIP/2.0
Supported: 100rel,path
User-Agent: OmniPCX Enterprise R10.1 j2.501.16.c
To: sip:4321@hq2.labjtr.fr
From: sip:4321@hq2.labjtr.fr;tag=bfc35e619db3ff4f042097e7b390c30a
Contact: <sip:4321@172.27.145.122;transport=UDP>;expires=1800
Call-ID: 5a4750d9baf3b90dd125dccb899bf474@172.27.145.122
CSeq: 571892426 REGISTER
Via: SIP/2.0/UDP 172.27.145.122;branch=z9hG4bK8d42eea8f1c72df626c86ea191f7ff76
Max-Forwards: 70
Content-Length: 0

Thu Nov 15 11:45:50 2012 SEND MESSAGE TO NETWORK (172.27.145.122:5060 [UDP]) (BUFF LEN = 396)
----------------------utf8----------------------
SIP/2.0 200 OK
Contact: <sip:4321@172.27.145.122;transport=UDP>;expires=1800
To: sip:4321@hq2.labjtr.fr;tag=2810b4ed27aa41ba89b99ef3631a8c0d
From: sip:4321@hq2.labjtr.fr;tag=bfc35e619db3ff4f042097e7b390c30a
Call-ID: 5a4750d9baf3b90dd125dccb899bf474@172.27.145.122
CSeq: 571892426 REGISTER
Via: SIP/2.0/UDP 172.27.145.122;branch=z9hG4bK8d42eea8f1c72df626c86ea191f7ff76
Content-Length: 0
```

11.12.10 Incoming call doesn't transit via Trunk Group configured on SIP Ext Gw

Symptom: When we make a trkvisu of SIP Trunk Group used by SIP External Gateway during an incoming call, we observed that no SIP Access is used.

Diagnosis: - by checking INVITE request received from Network, we can see that domain contained on FROM header is not recognized by SIP External Gateway, so call transits through Main SIP Gateway.

```
1332292333 -> Wed Mar 21 02:12:13 2012 RECEIVE MESSAGE FROM NETWORK (172.27.138.36:5060 [UDP])
----------------------utf8----------------------
INVITE sip:11001@172.27.144.20 SIP/2.0
Via: SIP/2.0/UDP 172.27.138.36:5060;branch=z9hG4bK15ac35dc;rport
Max-Forwards: 70
From: "Boss Hoggs" <sip:0033XXXXXXXXX@172.27.144.20>;tag=as5ff02451
To: <sip:11001@172.27.144.20>

Wed Mar 21 02:12:13 2012 [isDomainFromGwExt] Host from request is : 172.27.144.20.
Wed Mar 21 02:12:13 2012 [isDomainFromGwExt] User from request is : 0033XXXXXXXXX
Wed Mar 21 02:12:13 2012 [domain not from an External Gateway.

Wed Mar 21 02:12:13 2012 11cd[CMotorCall::onReceiveRequest] system option=0 extGw=-1.
Wed Mar 21 02:12:13 2012 11cd[CMotorCall::toGatewayOrProxy] request for proxydomain=172.27.144.20.
```

Solution: Modify FROM header sent by external application in order to match with remote domain configured on SIP External Gateway

11.12.11 Wrong caller number sent in case of forward

Symptom: Wrong caller number on OpenTouch anymobile device when using multi device feature.
Example: External user 0980406562 (phone A)
OT MIC SIP directory number 7905 (358306667905) (phone B)
OT anymobile number +358 (0) 505307947 (phone C)
Phone A calls phone B with a redirection to phone C. During phone C ringing phase, Calling Number of phone B is displayed instead of Calling number of phone A

Diagnosis: - Check if history-info/diversion header is present on requests received from OpenTouch with related forward informations
- Check External Signalling Parameters (Calling Name Presentation, NPD for external forward

Solution: NPD for external forward is configured at -1 so OXE sends redirecting number in case of forward. When parameters is configured with NPD used by SIP Trunk Group, initial Calling Number is sent.

```
Before NPD modification:
P-Asserted-Identity: "0501636" <sip:+358306667905@62.237.35.164;user=phone>
Content-Type: application/sdp
To: <sip:0505307947@194.100.41.52;user=phone>
From: "0501636" <sip:+358306667905@62.237.35.164;user=phone>;tag=77b6c1402197fc477d9268f1a0563007
Contact: <sip:+358306667905@62.237.35.164;transport=UDP>

After NPD modification:
P-Asserted-Identity: "0501636" <sip:+0501636@62.237.35.164;user=phone>
Content-Type: application/sdp
To: <sip:0505307947@194.100.41.52;user=phone>
From: "0501636" <sip:0501636@62.237.35.164;user=phone>;tag=10067c3f78682c28d55da5b1cc350f86
Contact: <sip:0501636@62.237.35.164;transport=UDP>
```

11.12.12 Diversion/History-Info header is not present

Symptom: User A (+33298285305) calls user B (1481001) located on PBX. User B is on immediate forward to User C (+33675445566). Second leg transits via the Trunk Group 16 (SIP ISDN All Countries) and SIP External Gw 2 (Remote domain: 172.44.266.44). Diversion header is not added by OXE.

Diagnosis: - Check External Signalling Parameters, Trunk Group and SIP External Gateway configuration

Solution: Configure following parameters:
System > Other System Param > External Signalling Parameters

NPD for external forward: 10 (NPD used by SIP ISDN Trunk Group)

Trunk Groups > Trunk Group
IE External Forward: Diverting leg information

SIP > SIP Ext GW
Diversion Info to provide via: Diversion

```
(013064:000323) | Diversion :
(013064:000324) |        Url : <> +332675445566@6.1.48.1
(013064:000325) |        Reason : UNCONDITIONAL
```

11.12.13 SIP-Trunking Name is displayed on calling phone set when call is established

Symptom: SIP Trunking Name is displayed on calling phone set when call is established with an external user through SIP Externl Gateway. SIP Trunk type is ISDN ALL COUNTRIES. Example: A is an internal phone set and dials external number +33014596222, when call is established, phone set doesn't display called number

Diagnosis: Check if SIP Carrier sends a P-Asserted-Identity header on SIP 200 OK Response when call is established.

Solution: Symptom: If no Called information is present on connection message (SIP 200 OK), OXE by default displays the trunk group name.

11.12.14 From header has not the national format

Symptom: Bad tagging of the calling from a SIP ISDN gateway

Diagnosis: When value on From header is not canonical, OXE tags the calling number like RNIS unknown

Solution: Modify the from received on OXE by adding canonical form and manage the country code like this the calling number will be tagged as national

11.12.15 Incoming and outgoing fax communications impossible through SIP Gw

Symptom: Re-INVITE with T38 on SDP is not sent by FAX Server, voice communication is cut before T38 négotiation

Diagnosis: As PBX is configured in spatial redundancy, FQDN is used. In this case, FQDN corresponds to the nodename concatenate with the DNS local domain name managed on SIP Gw. When OXE makes a fax call to Fax Server, FQDN is used on CONTACT header and as Fax Server cannot resolve it, call is cut.

Solution: Use an external DNS server for FQDN resolution or check at false the "Contact with IP Address" parameter on SIP Ext Gw.

11.12.16 No Re-Invite with T38 offer sent by OXE

Symptom: No T38 bascul during fax communication between PBX and FAX Gw

Diagnosis: On INVITE sent by the FAX Gw, FROM header contains the IP Address of PBX instead of IP Address of FAX Gw. So, when a Fax call arrives, this is the internal Sip Gw on PBX that is used and SIP-ABCF trunk group associated. RE-INVITE(T38) is only available on trunk group SIP ISDN.

Solution: Modify the IP Address on From Header sent by Fax Gw

11.12.17 External call with secret identity over SIP Provider fails

Symptom: Impossible to receive incoming calls with the secret ID

Diagnosis: When a call is received with the secret ID, the call is rejected by OXE with a 480 (not able to reach the third party)

Solution: The OXE is using the FROM field for the SIP gateway selection, in case of secret id, the FROM field contains this: anonymous@anonymous.invalid, so an external SIP gateway should correspond to the domain part of the URI, in that case anonymous.invalid (SIP Remote domain), this external SIP gateway has the same configuration than the one used to reach the SIP provider.

11.12.18 On SIP outgoing call, dynamic ports are used instead of port 5060

Symptom: why the OXE uses one of the dynamic ports for a SIP call instead of the port 5060?

Diagnosis: When a SIP trace is done with “wireshark”, the source port, when the OXE is the initiator of the call, can be different than 5060 (SIP port manage on the database)

Solution: Regarding the RFC3581, the initiator of the SIP call can choose a port number different than the default “SIP port” (5060) for its source port. So in that case the OXE is able to choose one port from the range of dynamic ports.

The important impacts about this behavior, it is the management of the size of dynamic ports range and also to take into accounts the configuration of the firewalls from the customer‘s network, to authorize them to use the dynamic ports for SIP communication.

11.12.19 A "+" character is added on calling number when ISDN call is routed to SIP

Diagnosis: Addition of "+" is normal, because incoming call from ISDN is tagged with 21 81 which corresponds to a National Call and according to the RFC, a “+” must be added before the Calling Number

```
| (033539:000002) Concatenated-Physical-Event :
| long: 40  desti: 0  source: 0  cryst: 1  cpl: 6  us: 0  term: 0  type a5
| tei: 0  >>>>  message received : SETUP [05]    Call ref :  00 37
|
|
| IE:[04] BEARER_CAPABILITY (l=3) 80 90 a3
| IE:[18] CHANNEL (l=3)  a9 83 8c -> T2 : B channel 12 exclusive
| IE:[6c] CALLING_NUMBER (l=6)  ->  21  81  Num : 2000
| IE:[7d] HLC (l=2) 91 81
|
```

Solution: The "+" is added because the calling party is tagged "national" on the ISDN call, so the OXE ia added the "+". None configuration must be done on OXE side.

11.12.20 Diversion Field has not the canonical form

Symptom: User A (+33298285305) calls user B (1481001) located on PBX. User B is on immediate forward to User C (+33675445566). Second leg transits via the Trunk Group 16 (SIP ISDN All Countries) and SIP External Gw 2 (Remote domain: 172.44.266.44). Diversion field has not the canonical form: 1481001

Diagnosis: Check NPD configuration, Diversion filed should be as follow: +331481001(canonical format) corresponds to +33 (France Country Code) 1481001 (Forwarded device number)

Solution: Configure a NPD for normal calls and a NPD for forward as below:

Here is NPD for normal calls:

```
Consult/Modify: Numbering Plan Description (NPD)

        Node Number (reserved) : 1
          Instance (reserved) : 1
          Instance (reserved) : 1
        Description identifier : 100

                    Name : SIP
```

```
Calling Numbering plan ident. + NPI/TON Isdn National
Called numbering plan ident. + NPI/TON : Isdn Unknown
Authorize personal calling num use + True
Install. number source + NPD source
Default number source + None used
Called DID identifier : 10
Calling/Connected DID identifier : -1
Installation number : 9839
```

And this is NPD for fwd calls:

```
Consult/Modify: Numbering Plan Description (NPD)

Node Number (reserved) : 1
Instance (reserved) : 1
Instance (reserved) : 1
Description identifier : 69

Name : FWD
Calling Numbering plan ident. + Unknown
Called numbering plan ident. + Unknown
Authorize personal calling num use + False
Install. number source + None used
Default number source + None used
Called DID identifier : 10
Calling/Connected DID identifier : 10
```

11.12.21 Leg1 and leg2 are external set, when OXE user performs a blind transfer, it doesn't work

Symptom: External UserA calls OXE user B thru public SIP Trunk(OXE user DDI: 210457060).
User B calls C (mobile phone) through public SIP trunk
B transfers the call to A before C answers
C answers the call but is not able to talk to external user, transfer is not complete by OXE

Diagnosis: Parameter "Support Re-Invite without SDP" is checked at TRUE on SIP External Gateway. Consequence is OXE doesn't perform transfer due to a R&D restriction on support of PRACK by remote according to this OXE configuration.

Solution: When PRACK is supported by SIP Provider, the parameter "Support Re-Invite without SDP" must be checked at false on SIP External Gateway.

11.12.22 SingleStep Transfer with REFER, no referred-by in the following INVITE

Symptom: OXE user A makes a call to an external SIP Server user B through SIP ABC-F Trunk. SIP Server user B makes a single step transfer to SIP Server user C with REFER method. In the following INVITE sent by OXE, the header referred-by is missing (see RFC 3892)

Solution: Since 10.1 (J2.501.21 release), a new parameter is available on System > Other System Param > SIP Parameters > Transfer : Refer using single step. This paramter is set by default at True and to obtain Referred-by in such case, it must be checked at False.

11.13 Summary for SIP issue analyse

The purpose of this chapter is give a way to analyse a SIP issue.

In case of SIP issue, a minimum of traces must be done, the “motortrace” trace is the minimum to make. The Infocollect must be done every time in case of SIP issue to get all the information needed to troubleshoot.

Here the different steps to start the analyse:

- Check if the SIP equipment is validated by Alcatel-Lucent.
- Check if the OXE configuration and SIP equipments respect the rules given on this document.
- Check if the CPUs belong to the IP domain 0.
- Check the “Network” management.
- Check the local SIP configuration (motortrace c).
- Check the incvisu file, and if SIP incidents, check the sipalarm files to find the causes of them.
- Check if an incident or a backtrace is generated when the issue is present.
- Check if the problem is from the SIPMOTOR or the Call Handling

If a SR will be openened:

- Provide a minimum of traces.
- Provide the call scenario (Caller, Called Party, IP addresses, etc...), provide all the information you can.
- Provide the Infocollect.
- Provide your anlayse of the issue, it is mandatory for you to make an anlyse before to open a SR.
- Provide a remote connection to the site (RMA, VPN, etc...)

BEFORE CALLING ALCATEL-LUCENT'S SUPPORT CENTER

Before calling Alcatel's Business Partner Support Centre (ABPSC), make sure that you have read through:

The Release Notes which lists features available, restrictions etc.

This chapter and completed the actions suggested for your system's problem.

Additionally, do the following and document the results so that the Alcatel Technical Support can better assist you:

Have any information that you gathered while troubleshooting the issue to this point available to provide to the TAC engineer (such as traces).

[Have a network diagram ready in case of ABC-F Networking problem].

[Have a data network diagram ready in case of VoIP problems. Make sure that relevant information is listed such as bandwidth of the links, equipments like firewalls, etc.].

[Have a VoIP Audit report available in case of VoIP problems].

Note
Dial-in access is also mandatory to help with effective problem resolution.

Comments
Adapt the paragraph if specific or additional information or actions are required depending on the subject.

End of document

Legal notice:
Alcatel, Lucent, Alcatel-Lucent and the Alcatel-Lucent logo are trademarks of Alcatel-Lucent.
All other trademarks are the property of their respective owners.
The information presented is subject to change without notice.
Alcatel-Lucent assumes no responsibility for inaccuracies contained herein.
Copyright © 2013 Alcatel-Lucent. All rights reserved